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Volume 11 - Number 4

RECORDING ENGINEER/PRODUCER

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. . . microphones for F.A.S., Maracana Stadium, Rio di Janeiro
. . . page 80

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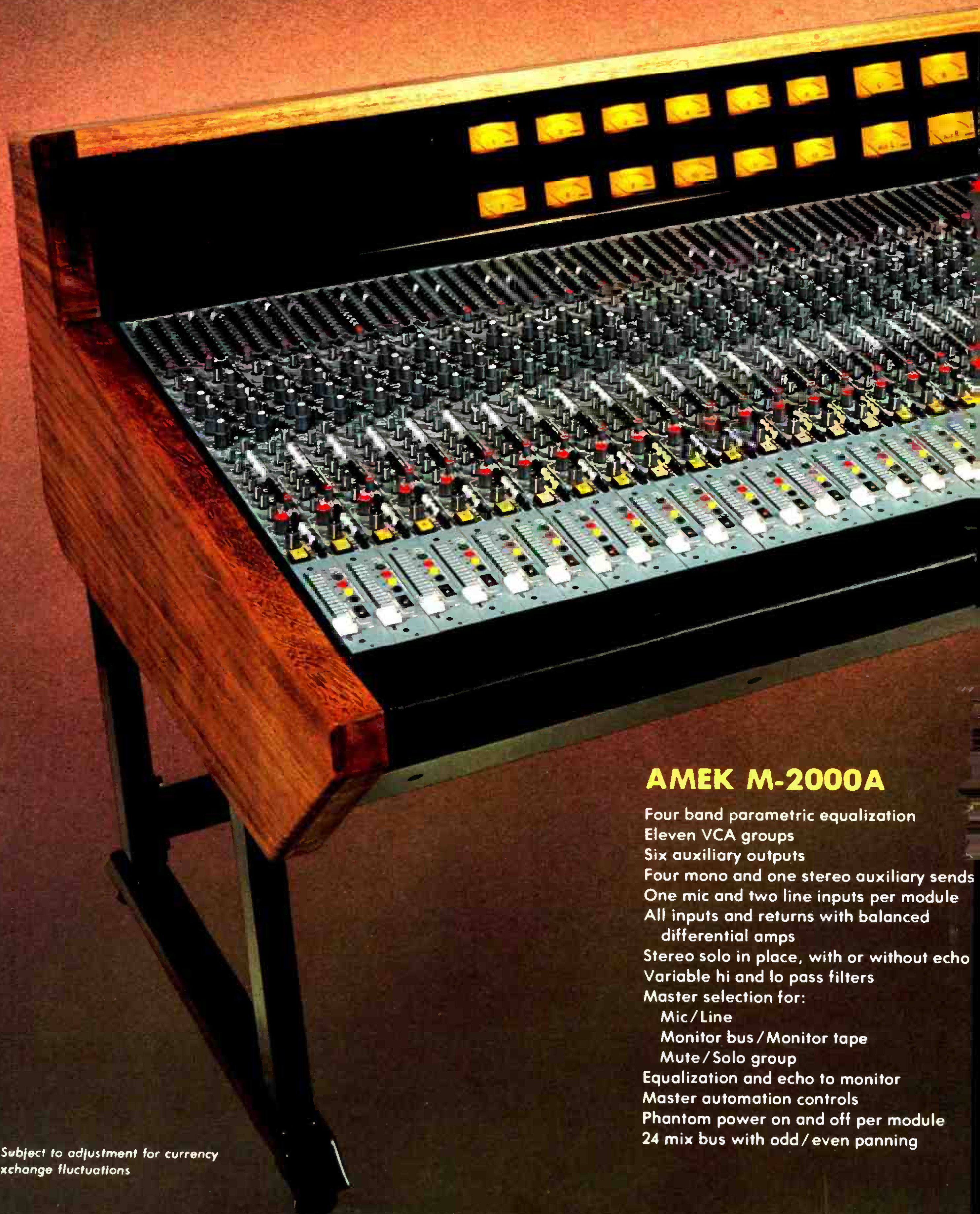

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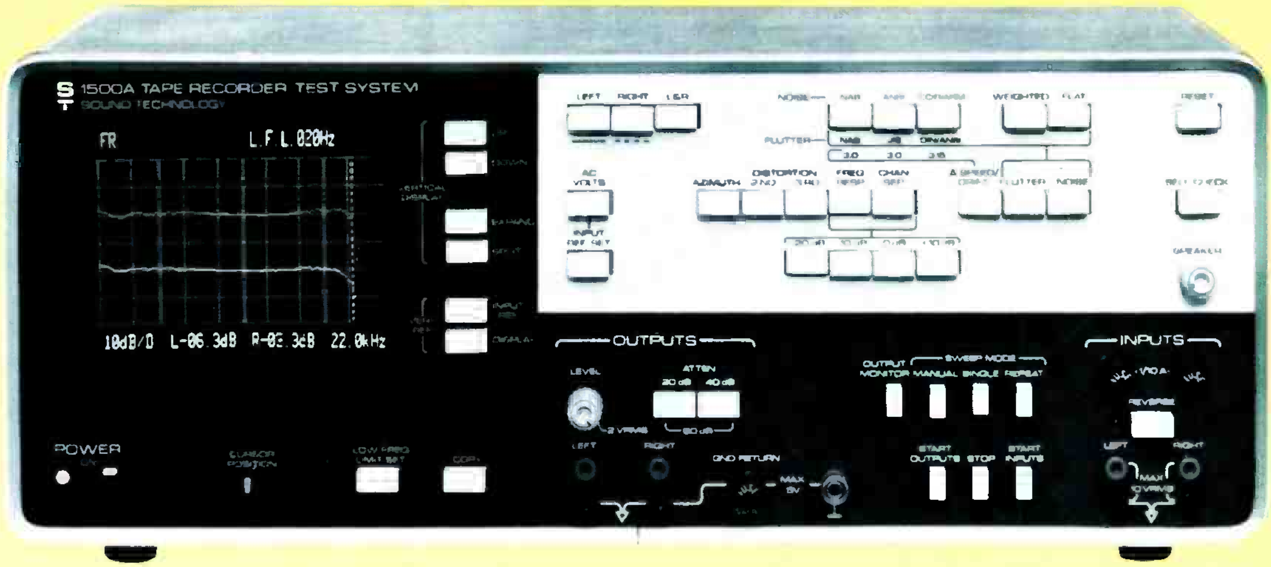

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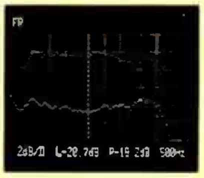
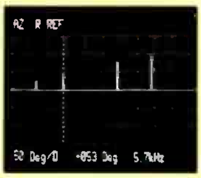


Two channel frequency response

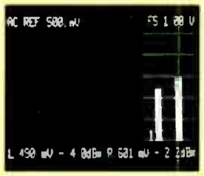
Third harmonic distortion vs. level

Flutter: 0.049% shown



Head azimuth accuracy

Noise: two channels; −53.4 dB shown



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views letters news

— U-47 CONVERSIONS —

from: **Stephen F. Temmer, President**
Gotham Audio Corporation
New York, NY

It is always unfortunate when statements are made in the press which simply are *untrue*. I specifically refer to the statement in Mr. Ethan Winer's "Letter to the Editor" *(June, 1980 R-e/p)*, that the capsule supplied as replacement in tube U47 Neumann microphones "is not the same as the one supplied with newer FET versions of the U47." I want to assure your readers that the statements which we have been making ever since the U47 FET was released are correct: Both the capsule and the head grille are *identical* to that used up to 1960 on the U47 models. Only the pattern switch, which allowed selection of cardioid/omni, and did no more than to connect the front and back membranes, was eliminated.

I would also like to comment that the long ago discontinued VF 14M tube which the U47s used, rarely if ever failed. I urge anyone who has such a tube in operation not to discard it even if he felt it had failed. Speak to Gotham instead. For anyone desiring the specifications for that VF 14M tube, they are available (in German) from us.

Gotham, as Neumann's U.S. representative is devoted to the knowledgeable maintenance of all Neumann's products, regardless of age. Bear in mind that the last U47 (tube) was built *20 years ago!*

Reply from Ethan Winer:

I can't understand why Mr. Temmer refuses to acknowledge that the original PVC capsules used in early versions of the U-47 were discontinued in later years and replaced with units made of Mylar. If I may quote from literature supplied by Gotham Audio (sheet #0469591): ". . . yours is a microphone below a certain serial number . . . made using a PVC plastic membrane which has hardened with time and now exhibits as much as 6 dB loss at 100 Hz referred to 1,000 Hz. The new element . . . has Mylar membranes and is not subject to this aging effect. *We are sure that you will find a major difference in the microphone's response quality . . ." (italics mine)*

Regarding the failure of VF-14M tubes, it is well known that Nuvistor conversions are performed on U-47s when the tube fails and a new tube either cannot be located or is deemed too expensive. If I may again quote from Gotham's own literature (sheet #0868i103): "It is to be bourne in mind that this conversion is NOT an improvement for these microphones, but rather a last ditch effort at keeping them in operation. DO NOT discard your VF-14M tube as long as it is working!" This is then followed by instructions for performing the conversion with step #2 reading: ". . . discard all six screws, 3 metal plates and the tube socket and tube."

I hope this clarifies any misunderstandings raised by my original letter on performing an FET conversion to a tube U-47.

Mr. Temmer's reply:

As to item one: He correctly quotes from one of our bulletins, whose number he also correctly quotes, and which shows it to be dated in 1969 (0469 .... means April 69), *long* before we ever thought of a U 47fet Solid State Unit. The entire bulletin deals with the change in membrane material FOR THE TUBE U 47 ONLY! His statement in his original letter which you printed, clearly stated that we were liars for claiming that the U 47fet had the same capsule as the U 47 tube after 1965; the first U 47fet appeared in 1972!

As to two: Our statement stands, that in our recollection we have never known a VF 14M Telefunken tube to have failed to the point of unusability. As was the custom during the tube era, people routinely replaced tubes in all equipment and many sent their mikes to us asking us to do just that. We had plenty of them and complied. We ourselves never installed any Nuvistor® kits, but simply sold them to people who insisted that their VF 14M had failed. We have never seen such failure. They did occasionally become microphonic, but proper shock mounting made them perfectly workable.

ed — Having spoken with Mr. Winer, **R-e/p** has been assured that there was absolutely no malicious intent in any of his publishings related to this matter.

— INDEPENDENT PRODUCTION —

from: **Will Connelly**
President/Producer
Star Jazz Records, Inc.
Fort Lauderdale, FL

Notwithstanding the critical comment by Gregory McKay *(R-e/p, June 1980)*, Dave Pell's article on *'Rules' for Playing the Independent Production Game, (R-e/p, April 1980)*, made a valuable and welcome insight into the contemporary nature of the business. The very fact that Pell's article drew comment is revealing, because it shows that there really aren't any fixed, uniform 'rules,' and I think his use of apostrophes to set off the word was fair enough warning that Dave knew that, too. The only consistent

— continued on page 14 . . .

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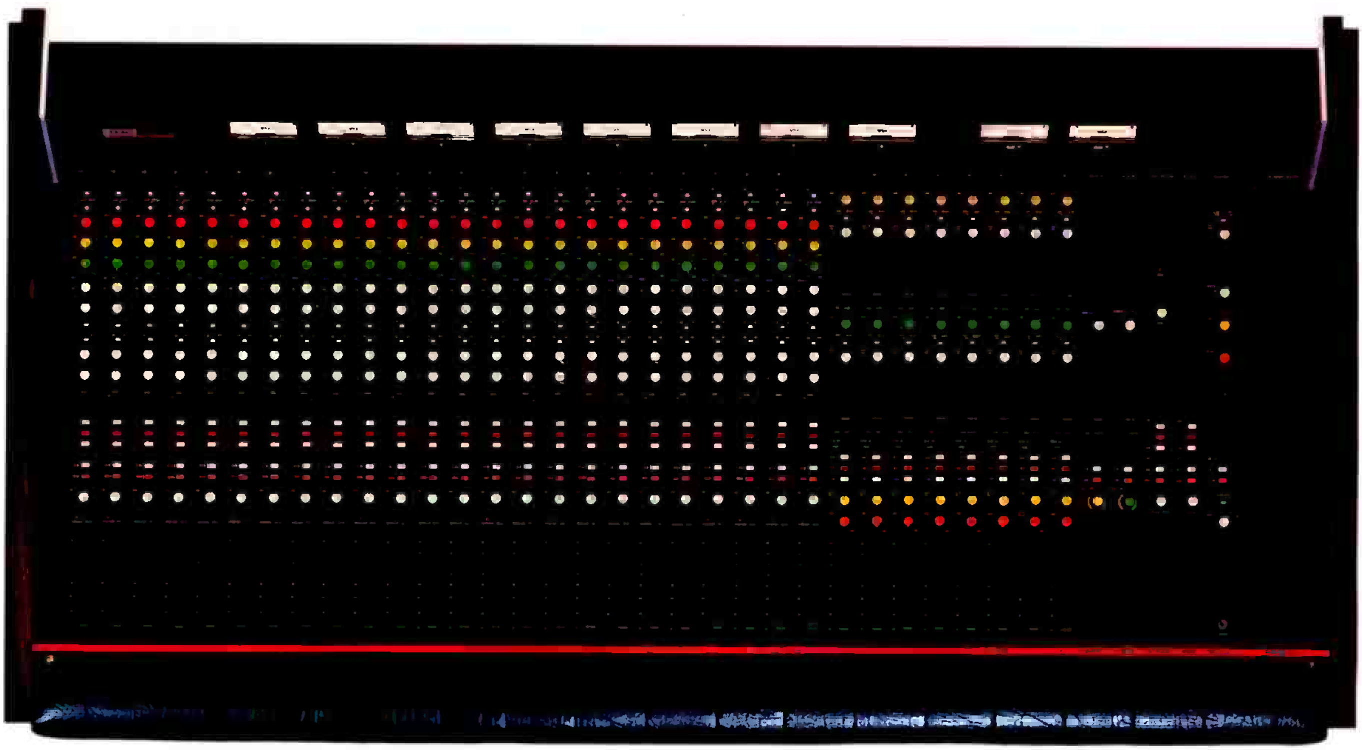

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But consider the Model 15. Rear panel patch points are already wired. Included in the cost. The meter bridge is already wired. Included in the cost. The separate power supply plugs right in. Also included in the cost. It's not unusual to get your board in the morning and do your first session that same night.

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. . . continued from page 8 —

rule of modern business practice that I am aware of is that every deal is a special deal, one that has to be (and is) structured within a framework of conflicting forces: desire, greed, tax factors, talent, reputation, material, and negotiating skill are but a few of the tangible and intangible factors that shape present-day music deals.

One thing is clear, and that is that Dave Pell wears many hats in addition to that of producer. It may be helpful to clarify just what it is that a producer does.

The producer is the creative decision maker. He may create the concept for a record himself, or he may evaluate concepts that others have developed, but the producer makes the final choice. The power (and responsibility) to make creative decisions extends to the choice of material, of artists, of arrangers, and to such elements as whether a 'sound' is right (e.g., use of reverb) and the order (sequence) of songs on the final record. The producer is the final arbiter on questions of musical performance. In exercising these sweeping decision-making powers, the producer is the *surrogate for the music-buying public*, and the producer's over-riding task is to create masters from which *marketable* records can be manufactured. Anyone can make records. A producer must make records that *sell*.

The producer is responsible for budget preparing. He must use his knowledge and experience to translate such factors as quality of musicianship into estimates of necessary session time and then into dollars. A budget is necessary so that adequate funds can be raised or committed to the production effort.

The producer is the manager of all the people who are involved in the making of a record: the musicians, arrangers, studio engineers and, indirectly, their subordinates. This is, of course, coupled with decision making. But decisions are meaningless unless, through diplomacy and persuasion, cajolery, threat, command, and any other means a producer may employ to motivate people to his will, the decisions are implemented. The crux is that once all the pre-production decisions are made, the producer has the duty and obligation to get from the first glimmer of the concept through to finished masters and to do so *within the budget that those who must foot the bills have approved.*

In none of the foregoing have I mentioned a producer's involvement in financing, promotion, publishing, copyright administration, and the like. This is because these are *not* producer functions. An individual who is a producer may very well get involved in all of these peripheral activities, but he does so in a capacity other than as a producer . . . such as personal manager or booking agency or corporate manager or financier or lawyer, etc. It will be helpful in future discussions to come up with similar "job descriptions" for other members of the record making team.

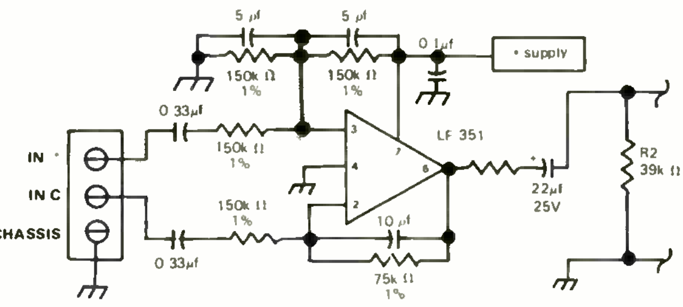

FIGURE 3
BALANCED TRANSFORMERLESS INPUT CIRCUIT
(Optional)

— CORRECTION —

In David Baskind's article, **Modifying the LA-3A,** which appeared in the June 1980 issue of ***R-e/p*** (Page 40), an omission occurred in the drawing of Figure 3. An ellipse indicates where the omission was corrected in the re-drawn circuit.

What The
AUDIO/VIDEO fusion
Can Mean To The
RECORDING STUDIO INDUSTRY

— series conclusion —

LIGHTING & MAKEUP for producing VIDEO DEMOS

by Steve Barnett

In recent issues of *R-e/p*, we have discussed what is being called the Audio/Video Fusion. The explosion of cable television and home video recorders, and the advent of the video disk have posed questions for the recording industry with regard to video music, the visual interpretation of an artist's musical work. In the first of this series, we examined the potential marketplace and the intricacies and costs of high end video production. We then explored lower levels of video production currently being undertaken, and in the last article, we offered a sampling of video equipment available in that price range should a studio owner wish to become involved with actual video production. In this final piece in the series, we discuss basic

— continued overleaf . . .

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lighting and make-up for the beginning video producer.

Lighting

"One of the factors that makes video music demos kind of tricky," says Doug Jeffs, (Hoffman Video, L.A.) "is that rock bands usually perform under low light conditions. In order to get a good readable picture from a color camera in low light you're looking at broadcast quality cameras ($20,000 each and up). These cameras, (the Hitachi GP7A) will operate down to about 10 or 20 foot candles, but generally the thing that's tricky is the color registration. You can get a readable picture, but you might end up with skin or flesh tones that look a little purple or a little green and in those situations you don't have enough light."

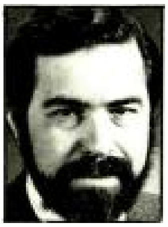
"The cameras out there today are doing a surprisingly good job," says Polon, "but one thing to remember about cameras today is that they're a lot like shortwave radios. The thing that determines the quality of your signal pick-up is not so much the radio as it is the antenna. With color cameras, it isn't so much the camera as it is the lighting. How well is the stage lit?

"If you're going to go into it, you have to make sure you've got decent lighting, otherwise the best camera cannot turn out a decent picture. At the networks, the picture quality is handled by the lighting director, the lighting console operator and the video control operator. The picture quality is definitely a 50-50 participation between lighting and video."

In the February issue of R-e/p, Bill Klages, a lighting designer for Imero Fiorentino Associates, commented on the difficulty of lighting a live performance for video. "In a concert situation," he explained, "the lights are usually designed for the benefit of the audience in attendance. Hence, when you're taping a concert, the lights must be altered to get an acceptable video picture.

"Intensity is a problem,and the camera can only see a certain number of things, so if you start to mix too many colors, it comes out white, or yellow, or some other funny color. What must be remembered is the eye can see a great deal more variations of hues and shades than can a camera. Generally the end result is only fair unless you start all over again and make it specifically for television. The two do not really mix."

It should be noted here that there are video producers and directors working in club situations with available stage lighting who prefer the gritty results achieved, feeling that it lends itself to the rock music being performed. Should one feel the need for enhancement, however, additional lighting instruments may be required. Thomas



Pincu, vice-president of marketing for Colortran Lighting in Burbank, California, addresses the problem of lighting for these small video production packages. "If you need a portable lighting package to handle a four or five piece group, then we have a kit that's pretty sizable, called the production kit," he says. "It has a total weight of 176 pounds and comes in

two carrying cases with quite a variety of equipment and has a retail price of around $2,450. It's complete with spotlights and floodlights as well as stands, extension cables, and distribution boxes for every fixture.

"The production package contains three 1,000-watt Fresnels, two 1,000-watt broad floodlights, which are fixed focus, and a 500-watt multi-focus broad flood, and they all come with their own diffusion frames and barndoors."

The Fresnel instruments are focusing lights, usually used as the key light on a subject. The fills are a bit softer and serve, usually as back or fill lighting. Barndoors are placed on the lighting instrument and adjusted to keep light off a particular area, while the diffusion filters are placed in the frames of the lights to reduce the intensity of the beam, measured in foot candles, without reducing the color temperature, read in degrees Kelvin. The color temperature is not related to light intensity per se. Rather it is the color of a given light as registered by a film or video tape medium. Daylight for instance has a Kelvin temperature of roughly 5,200 degrees, and reads in the bluish scale on film that's been balanced for indoors. Tungsten or artificial light generally reads about 3,200 degrees on the Kelvin scale, and will cast an orange tinge on outdoor balanced film, or on film in a camera that's using a filter to balance the indoor stock for sunlight.

The nuances of video lighting have been written about extensively and fill many books. It cannot be thoroughly explained in a few brief paragraphs as audio engineering cannot. However, there are some basics that should be defined. One of these is depth of field, or the area of acceptable focus. If you focus sharply on a subject before the camera, an area roughly one-third before the subject and two-thirds in back of the subject will be in acceptable focus. Background and foreground will gradually diminish in clarity the farther in front or to the rear an object is in relation to the point of principle focus.

Depth of field is affected by, among other factors, the f-stops in the camera lens. This scale is marked in numbers sequenced 1.8, 2.8, 4, 5.6, 8, 11, 16, and 22. The smaller the number, the wider the aperture of the lens and the shorter the depth of field. That inverse relationship can be confusing and should be double-checked. The f scale is also square in nature. An aperture of f-4 will allow twice as much light to pass through the lens as will f-5.6, and f-11 will allow ½ as much light to pass through the lens as f-8.

The smaller the aperture (or the higher the number on the F scale) the greater the depth of field; or put another way, the area of acceptable focus of a picture taken through a lens with a f-16 or f-22 setting will have a background that seems to be nearly as sharp as the point of principle focus. Alternately, the larger the aperture or the lower the number on the f scale the smaller the depth of field. For instance, f-2.8 will produce a picture that has the subject in sharp focus, with the background and the foreground blurry.

Other factors affect depth of field, such as the focal length of the lens and the distance from the point of principle focus, but these can be looked at in greater detail in any number of books on basic photography.

The reason it is important to be aware of depth of field and its relationship to light becomes apparent when video tape recording is undertaken with live performance lighting conditions. In order to achieve a readable picture on the video screen it may be necessary to open the aperture of the camera to a wide open setting, say of 2.8, thus affecting the depth of field. The area behind the lead singer will be out of focus, depriving the video of a clear view of the band behind the singer. To achieve a greater depth of field, more light must be added to the stage area in order to be able to stop down, or close the apertures of the cameras. On the other hand, you may want the background out of focus, to hold the audience's attention on a point in the foreground, say a guitarist involved in a solo on the edge of the stage.

"Take a look at the dramatic productions that CBS Television did years ago," adds Pincu, who received a Masters Degree in Television from UCLA. "They used very low lighting levels and operated their cameras extremely wide open f-3.5 or -4. They had extraordinarily limited depth of field so that the director could determine exactly what it was that he wanted the viewer to see."

"NBC and ABC on the other hand, doing the exact same kind of work, tended to light at higher levels. They would run at f-8, f-11, or f-16 on their cameras. Since they illuminated at those higher levels, they could stop down farther, and they could get increased depth of field. In variety shows, like Dinah Shore, they'd run at f-8, and have a depth of field of 20 feet."

"There were all kinds of schools of thought. I preferred the selective focus technique that CBS used. It tends to be more cost effective, it's more comfortable to work under (less light, less heat), and the director permits the viewers to see only what he wants them to see sharply."

A number of photography manuals have charts and graphs to help you plan your depth of field with regard to all the factors mentioned. In the end, however, the creative decision belongs to the video director.

"For additional flexibility in lighting a band," Pincu suggests that "two more Fresnels be added to the lighting kit.

— continued overleaf . . .

What The
AUDIO/VIDEO fusion
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Particularly if you are going to be lighting a stage with five to six people on it."

This would allow the use of three Fresnels for key lighting on the most important subjects, two Fresnels for back lighting, and the three floods for fill. In lighting remember that definition is achieved through the skillful use of shadows, by having one side of a performers face a little lighter than the other. This helps create the illusion of depth and eliminates that flat look so often associated with wash lit variety shows.

In addition, says Pincu, "For the guy who's just going out on the fly and wants to know what he's got, I would definitely recommend a good incident light meter."

Using the camera's specifications, then, the lighting director can plan the ratios of light to shadows, of key to fill, or back light to fill, to give the lighting depth and texture.

"I think that a basic understanding of photographic technique is clearly important," continues Pincu. "I got my greatest appreciation for light and how it works when I started working with a 35 millimeter still camera. By pressing the preview button while looking through a single lens reflex camera, I was able to instantly see the effect of depth of field and selective focus. That's when I really began to appreciate it. When I began lighting shows for television, I feel I was better able to do the job. I had a much better feel for it. I think it's extremely important that you understand the technical aspects of lenses and light."

It must be underscored at this point that these are only the most basic principles of elementary lighting design, and one cannot possibly learn this incredibly complex and technical art by simply reading an article such as this, or even a massive volume on the subject. The artistry involved in broadcast quality video lighting takes years of study and experience to master, and in many situations, it will take more lighting fixtures to attain this high quality picture, and this will not fit into the budgets we have been dealing with here.

"Even though you may not be using it for broadcast," offers Bob Slutsky, of Skirpan Lighting, "you have to remember that everybody is looking at it with broadcast quality lighting in mind. That's how the tape and ultimately the group will be judged."

He agrees with Klages that rock theatrical lighting is extremely difficult to mix with the lighting demanded by video, but compromises must be made in order to get an acceptable picture. The problems are not only of camera gain and video quality, but of conflicting theories of lighting a musical performance.

"John Denver will have none of being lit two different ways," says Slutsky. "He feels that the audience in front of him is the most important audience, so you must keep the theatrical lighting, yet you still must get enough illumination on the performer.

"A good video operator and a good lighting designer can think their way through this, but it is very complex due to all the variables involved, and you can't pull it off if you don't know video."

At this stage, Slutsky expresses doubts as to the value of such a video package if the budget limitations restrict the number of lights available, and if the experimenter in video lacks the knowledge of the video chain required to get and keep a quality picture.

"Anybody can get a picture," he says, "but to get a picture that's of acceptable quality is what separates the men from the boys, and to top it off, every situation will be different. You will never light two groups the same way, so because you never know what you're going to encounter, you better know what you're doing."

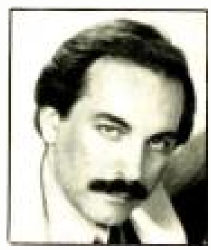
Make-Up

Hand in hand with the question of lighting is the concern for make-up for the performers.

"Make-up is important," agrees Polon. "Again, how many people look is both a function of both how they're lit and how they're made-up. You always want to try to show people at their best, and make-up helps do that, because cameras can accentuate features which may not be desirable to highlight. This is true of all cameras, not just video cameras.

"Skilled use of make-up is a common tool to help create the illusions we're seeking. Let's be realistic. Make-up isn't cheating anymore than multi-track audio recording is cheating. We're using technology to enhance the illusion of that performance. The same thing holds true for video.

"It's not fair to put someone on a TV show when they've just come off of two weeks of the Hong Kong flu and look like death warmed over. But if that's when they have to go on the tube, they've got to look their best. Make-up can go a long way to help."



Robert Salvatore is Beauty Director for Max Factor in Hollywood, and has been involved with Sony Corporation in studying make-up for semi-pro video venues.

"What we discussed with the Sony people," he explained, "was that most people are not make-up artists nor are they used to wearing make-up, and yet an increasing number are having to deal with it. This is the obvious result of the expansion of visual mediums, and it's very important.

"Sony brought in the sets and the lighting conditions that would exist (in industrial video), and unlike most television where people normally look washed out and you compensate with darker makeup, I discovered that with these small home or executive taping systems, you actually look darker instead of ligher. I suppose it's because of a lack of lighting or something.

"The thing to tell a woman involved with this kind of thing is that she should just wear the make-up that she wears everyday,

— *continued on page 126 . . .*

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STUDIO UPDATE

Northeast:

■ **THE INSTITUTE OF AUDIO RESEARCH** (New York City), calling itself the largest school for multitrack recording technology training in the country, is currently expanding its 16-track in-house training facility. Completely equipped for hands-on console, tape machine and outboard equipment use, the facility will include a performing studio designed by architect **JOHN STORYK.** According to executive director, **PHILLIP STEIN,** Storyk transformed a relatively small room into an effective, functioning, acoustically treated sound studio which will be used to train music students. **ALBERT GRUNDY,** president of the twelve year old institute, has announced that the studio will be completed by September.

■ **RED GATE STUDIO** (Kent, New York) co-owned by musician/producer/record company owner **GENE PERLA,** and musician/producer **JAN HAMMER,** have recently upgraded their studio to 16-tracks featuring the MCI JH-24 with AutoLocator and a Sound Workshop Series 30 console, 20 x 16, format "B", with VCA grouping. Artists recently recording at Red Gate include **JEFF BECK, STONE ALLIANCE, JAN HAMMER, ELVIN JONES** and **TONY WILLIAMS.** *c/o P.M. Records, Inc., 20 Martha Street, Woodcliff Lake, NJ 07675. (201) 391-2486.*


Soundesigns

■ **SOUND IDEAS STUDIOS** (New York City) has become the first studio in that city to receive a 3M digital mastering system, consisting of 4-track and 32-track recorders. According to owner **GEORGE KLABIN,** the acquisition will provide greater versatility to the artist. Sound Ideas' digital system can be used in either newly re-built Studio A or in Studio C, which can handle up to 40 musicians. **BOB SCHAFFNER,** studio manager, says that musicians he has talked to feel that digital playback sounds just like the performance. He believes that digital recording is the next logical step to increase attention to room acoustics and ambience as opposed to electronic equalization. One of the first groups to test the multitrack system at Sound Ideas was the **BT EXPRESS,** a rhythm-and-blues instrumental group recording for Columbia Records (producer **MORRIS BROWN**). Rhythm-and-blues, jazz, and commercial jingles are Sound Ideas' primary productions. Sound Ideas also offers 3M's electronic digital editing system and a digital preview unit. The studio is actively demonstrating the system to interested parties and is currently offering digital recording at competitive analog rates. *151 West 46th Street, New York, NY 10036. (212) 575-1711.*

■ **STARR RECORDING, INC.,** (Philadelphia, Pennsylvania) announces that **CARL PARUOLO** will take the position of chief engineer and studio manager of the 24-track facility. Paruolo was formerly studio manager and then chief engineer at Sigma Sound Studios. The appointment was made by Starr president **DAVID STAROBIN.** *201 Saint James Place, Philadelphia, PA 19106. (215) 925-5265.*

Southeast:

■ **SANBORN PRODUCTIONS** (Nashville, Tennessee) has completed construction of Studio A at their Bull Run Studios located on a 28-acre estate on the Cumberland River. The room, designed by **JOHN GARDNER** and **CARL FROST,** features large picture windows and a wall built entirely out of quarry stone. Studio president Frost also notes that Sanborn's Mobile One has installed new outboard gear, including an Eventide Harmonizer and flanger, a Scamp rack with compressors, gates, and EQ, an Orange County Vocal Stressor, and a MICMIX XL 305 reverb. *Box 120, Route 3, Ashland City, TN 37015. (615) 254-6538.*

■ **OPRYLAND PRODUCTIONS** (Nashville, Tennessee) has announced the opening of their new Audio/Video Editing Suite, featuring a new Soundcraft 1624 console, Ampex 2-, 4- and 16-track machines, a full audience effects library, Marshall Time Modulator, EMT and AKG echo, Quantel, and Ampex video effects, plus EECO and CMX 340X sync and editing facilities for multitracks, 2" and 1" VTRs in any format. *2806 Opryland Drive, Nashville, TN 37214. (615) 889-6840.*


. . . the Back Room

Midwest:

■ **MURRAY ALLEN,** president of **UNIVERSAL RECORDING CORPORATION,** Chicago, Illinois, has announced the unveiling of **THE BACK ROOM** recording studio, the only facility in the Midwest specially designed to handle audio sweetening for video. In addition to the audio-to-video mixing capability, the studio features 24/48-track recording with Ampex 2-, 4- and 24-track machines as standard equipment. An automated MCI 600 series console, Sonly video tape recorder, BTX video computer, and UREI 813 speakers are a few highlights from a long list of top-flight equipment. In so much as The Back Room is Universal Recording's future, it is also very much a part of its present. Booked solid for two months, the studio has already been discovered by commercial music producers and recording artists alike. *46 East Walton Street, Chicago, IL 60611. (312) 642-0665.*

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these are some of the interesting news items that can be announced in the next available issue. Write:
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STUDIO UPDATE

■ **RED SKY STUDIO** (Steger, Illinois) has added to its mike collection six Shure SM-53s, two SM-76s, and an SM-81 condenser. Owner/operator **MICHAEL ICZKOWSKI** has also installed overhead cue sends and an RCA tube limiter to the 8-track facility. *3419 Sally Drive, Steger, IL 60475. (312) 754-6297.*

■ **STUDIO G RECORDING** (Clayton, Missouri) is a new facility expressly designed to accommodate commercial and audio/video production, motion picture post-production, and location recording. The equipment, provided by Milam Audio, includes a Neotek console, MCI JH-110-8B recorder, and a complete interlocked film dubbing system, with outboards by UREI, Technics, and Sound Workshop. Monitors are McIntosh-powered Big Reds. **GREG GLAZIER** is the owner and chief engineer. *214 South Bemiston, Clayton, MO 63105. (314) 727-0770.*

■ **THE CHICAGO RECORDING COMPANY** (Chicago, Illinois) has installed a new CADAC recording console and CADAC monitors in their operation. *528 N. Michigan, Chicago, IL 60611. (312) 822-9333.*

■ **A&R RECORDING** (Ames, Iowa) is operating in a newly remodeled facility which includes a new control room and an overhead client observation room. Studio owner **STEVE MONROE** and manager **DAVE KINGLAND** have also installed a Neotek Series III console to go with an MCI 24-track recorder. Twenty-four tracks of dbx and UREI Time Aligned™ monitors are also in use.

South Central:

■ **RAMPART RECORDING STUDIO** (Houston, Texas) has added **DANIEL YEANEY** to the engineering staff. Rampart is a 16-track facility featuring 3M tape machines, custom built monitors with Crown amps and crossovers. The outboard equipment includes an Eventide Harmonizer, UREI compressors, EMT stereo plate reverb, and assorted parametric and graphic equalizers. Also available are a Hammond B-3 and a grand piano. *6105 Jessamine Street, Houston, TX 77081. (713) 772-6939.*

■ **ACA RECORDING STUDIOS** (Houston, Texas) is a professional recording service offering two up-to-date facilities. Studio A is a 24 into 16 studio with an Auditronics board, 3M, Ampex, and Scully recorders, and dbx noise reduction, plus a myriad of outboard gear. Studio B is a 16 into 4 studio with an Auditronics board, Ampex, and Scully recorders, and dbx noise reduction, plus outboard gear. The head engineer is **BILL HOLFORD,** who has at least 32 years of recording behind him. Additionally, the studio boasts a new second engineer from Australia, **ANDY BRADLEY.** *8208 Westpark Drive, Houston, TX 77063.*

■ **SOUND ARTS RECORDING STUDIO** (Houston, Texas) has completed their new control room and lounge with patio and bar area. The studio features a new Speck 48-input console, and 16-, 8- and 2-track 3M mastering machines. **WOODY SMITH,** of ASI (San Antonio) was the consulting engineer for the design and the acoustics of the new room. Current projects include **BUDDY MILES** laying down tracks with Houston band **TEMPEST** — engineering by **JEFF WELLS,** assisted by **MIKE LATVELLA;** New Wave band, **THE RAKES,** from Virginia, have completed their album for **EARTH** Records International. Engineering by Jeff Wells, assisted by **GEORGE SCHWALM.** The tune, "Let's Get Drunk and Be Someone Tonight" (written for the "Urban Cowboy" movie), was recorded and released as a 45 on Earth Records. *2036 Pasket, Suite A, Houston, TX 77092. (713) 688-8067.*

Rocky Mountain

■ **THE LAST RECORDING STUDIO** (Boulder, Colorado) has successfully completed its first year in operation. The facility has upgraded to the use of Otari 2- and 4-track tape machines and is equipped to mix up to 16 channels. Microphone selection has expanded to the use of AKG, Sennheiser, Neumann, Electro-Voice, and Shure. All equipment is also available to do remote recordings, past clients include **THE RICK DANKO-PAUL BUTTERFIELD BAND** and jazzmen **RICHIE COLE** and **PHIL WOODS.** Future plans include continued expansion to comfortably meet the increasing needs of the recording community in Colorado. *Box 6050, Boulder, CO 80306. (303) 442-1158.*


Radiant Star

■ **RADIANT STAR STUDIOS** (Loveland, Colorado) has added an Ampex AG-440 one-inch 8-track recorder according to studio manager **BRUCE BRUNSON.** Radiant Star also offers a full complement of studio instruments including a grand piano, drum kit, and synthesizers. Cassette duplication services are also available. **DAVID WOLPERT** and **TONY ZOTTA** are staff engineers. *204 East Fourth Street, P. O. Box 192, Loveland, CO 80537. (303) 669-5912.*

Southern California:

■ **THE MUSIC GRINDER** (Los Angeles) has expanded by 2,700 square feet and is now renting rehearsal and showcase facilities. *7460 Melrose Avenue, Los Angeles, CA 90046. (213) 655-2996.*

■ **RECORD PLANT** (Los Angeles) has purchased a second 3M digital mastering system, consisting of 4-track and 32-track recorders equipped with cross-fade editing capability. Installation was scheduled for August. The studio was the first to receive a 3M multitrack digital system in February 1979, and during recent months has used a second, rented, system much of the time to meet recording demand. The new system is ultimately destined for Record Plant's totally new Studio D, which will open in January. This 48 x 52 foot, 65-person capacity studio, designed by Tom Hidley, will be equipped with state-of-the-art equipment for multi-media production, including film scoring. The studio also has a 3M digital editing system and will have a digital preview unit for record

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mastering. In addition to the digital mastering system, the new Studio D will be equipped with an SSL console (48 in, 32 out), a 24-track analog system and 35 mm projectors with magnetic transfer capability. "Film makers are keen on transferring mixed down multitrack digital sound direct to mag track," according to **CHRIS STONE,** Record Plant president. "We expect to be the first studio in Los Angeles especially designed to do this."

■ **CITY RECORDERS** (Hollywood, California) is expanding existing 24-track facilities and adding a new 16-track studio. Studio A has been rebuilt and opened the week of August 11 with a new Trident Series 80 console, 40 x 24. Also, there are UREI Time Aligned™ 813 monitors with MCI, Ampex, and Scully recorders. A total complement of microphones and the usual outboard gear and echos, both digital and plate, round out this room. Design is by City staff and installation and wiring by **STUDIO MAINTENANCE SERVICE.** Studio B at City is as big as A, 50' x 25', and the control room at 12' x 20' almost as big. A full stage in a bright room is the main feature here. The console is a 20 x 16 Eltec with Big Red monitors, Ampex, and Otari recorders. Echos and mikes and outboard are reservoired with Studio A. Both studios have their own 6', 8" Kawaii grand pianos. City Recorders is located in the heart of the east Hollywood record industry at Sunset Boulevard, yet it is totally isolated with 24-hour security and absolute privacy. The facility is also close to the best in restaurants and catering services. City presents a new huge, plush lounge complete with ON-TV, designed by **JAN ADAMSON,** of Lusk Interiors.

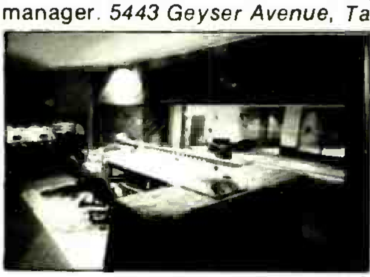
■ **REDONDO PACIFIC STUDIOS** (Redondo Beach, California) announces the completion of flush mounting of their custom UREI Time Aligned™ monitors. New decor is also in the works, according to **CATHARINA MASTERS.** *612 Meyer Lane, #18, Redondo Beach, CA 90278. (213) 652-9498.*

■ **THE INSTITUTE OF SOUND RECORDING** (San Diego, California) has opened its 24-track Studio One for use in its continuing recording engineering program as well as for public bookings. A Spectra Sonics console is employed with MCI 16- and 24-track machines, each with AutoLocator. Monitors are by Tannoy, JBL, and Auratone, while outboard equipment includes Lexicon Prime Time DDL, a dbx 162 stereo compressor, UREI 176 compressor/limiters, Orban parametric EQ, and a Sound Workshop vocal doubler. Mikes are by AKG, Sennheiser, Sony, and Shure. *8245 Ronson Road, Suite L, San Diego, CA. (714) 281-7744.*

■ **PARAMOUNT RECORDING STUDIOS** (Hollywood, California) has opened its new Studio C complex, constructed from the ground up. The new facility features a 28-foot Hidley-style control room, Studer machines, automated 48-track capability, UREI monitors, and a 45-foot studio. A nine-foot grand piano is also offered along with client offices, kitchen, and jacuzzi. *6245 Santa Monica Boulevard, Hollywood, CA 90038. (213) 461-3717.*

■ **UNITED WESTERN STUDIOS** (Hollywood, California) has appointed **CARY FISCHER** to the post of engineering manager. Fischer has been on United's maintenance staff for the past five years and is a native of New York City. The announcement was made by vice president and general manager **JERRY BARNES.** *6050 Sunset Boulevard, Hollywood, CA 90028. (213) 469-3983.*

■ **REDWING SOUND** (Tarzana, California) is a full format 24-track studio with a Trident Flexmix console linked to an MCI recorder. The board is transformerless as are the limiters, which include solid state models and tube units by Fairchild and Teletronics. Redwing also offers Lexicon Prime Time, the DeltaLabs Acousti-Computer, and the Parasound De-esser. Mikes are by AKG, Sony, Neumann, and Sennheiser; while the 25-by-30 studio features a Kawai 6-foot, 8-inch grand piano, two drum kits, and a number of synthesizers. **TOM SEUFERT** is Redwing's owner, while **KIRK BUTLER** is the studio manager. *5443 Geyser Avenue, Tarzana, CA 91356. (213) 344-5692.*


L.A. Studios

■ **L.A. STUDIOS** (Hollywood, California) has completed initial construction and is awaiting delivery of an MCI JH-600 console and an MCI JH-24 transformerless master recorder. The studio, which has been wired for video, is owned by **JOHN FRECHETTE**, of Pacific West Recorders of Redmond, Washington.

■ **THE CUTTING SYSTEM, INC.,** (Burbank, California) has been opened by **KEVIN GRAY** and **DOUG SHEPPARD** with all electronics, from tape head to cutting head, completely transformerless. A Neumann SX-74 cutting head operates a Neumann lathe with a custom pitch computer. Other gear includes Studer tape machines, Sontec four-band parametric EQ, and UREI 813 monitors. *3307 W. Magnolia Boulevard, Burbank, CA 91505. (213) 841-5884.*

Northern California:

■ **TEWKSBURY SOUND RECORDERS** (Richmond, California) and **RANCHO RIVERA RECORDING** (San Francisco) have acquired the studios at Hyde and Eddy Streets, in San Francisco, formerly Filmways/Heider Recording. The facilities include four complete studios and five acoustic echo chambers. The new studios will be known as **TEWKSBURY/THE HYDE STREET STUDIOS,** and will feature a 40 input Trident console and 28 input Helios console, and a new Otari MTR-90 24-track recorder as well as an Ampex multitrack recorder along with an extensive selection of signal processors and microphones. Rates will be substantially less than have been available before — presently quoted at $40.00 per hour for 24-track, and $25.00 per hour for 16-track. A minimum of ten hours is required for the above rates. The studios are available on a 24-hour basis. *245 Hyde Street, San Francisco, CA 94102. (415) 232-7933.*

■ **TEWKSBURY SOUND RECORDERS,** at their Richmond, California location, has completed installation of a new Helios console and a new pair of Tannoy Studio Gold monitors. Other additions of late include a Marshall Time Modulator, a Lang PEQ1 tube EQ, and a small capsule tube Sony condenser C-17b mike. *6026 Bernhard, Richmond, CA 94805. (415) 232-7933.*

England:

■ **ROCKSTAR STUDIO** (London, England) is approaching its third anniversary of operation and is now a 24-track facility. Its 12-by-12 foot control room features an Allen & Heath Syncon console feeding a 3M M-79 multitrack recorder and a Scully 280 mastering machine. Tannoy Gold monitors are utilized powered by Quad 303 amplifiers. Among the sideboards are a MICMIX Master Room spring reverb, Rebis noise gates, a Klark-Teknik DN34 analog time processor, and an Itam 3.77 tape deck for repeat echo and reverb. The recording room is 20-by-16 feet in size. **JOHN SPRINGATE** is the studio manager. *West End, Longdon, England.*

■ **1.2.3. MUSIC STUDIO** (Cardiff, Wales) has been in operation for several months handling radio and jingle production

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Recording Equipment: it's more than just hardware

by Thomas J. Valentino, Jr.

According to a recent survey, more than 72% of all recording studios utilize, as part of their services to clients, both a Production Music Library and Production Sound Effects Library. Now that figure might come as quite a shock to many people in the industry inasmuch as the primary function, one would imagine, of a recording studio is to record music and sounds originally for their clients. However, in these days of tight money, tighter budgets, and still tighter time schedules, many of the clients of recording studios are opting for the use of pre-recorded music and sound effects in their productions of TV and radio spot commercials, slide-programming, video productions, and films. So the recording studios need to keep on hand — for just such uses — complete Production Music Library and Sound Effects Libraries. In this way, the recording studio keeps its viability in the face of changing and specific needs. Let's take both items separately and see how they measure up in many areas for the recording studios.

The Production Music Library

The Production Music Library, as a species, is something totally unique. Best seen in the comparative sense of a "book" library, a good production music library is very much the same thing. Where a book library may have different categories of books, (i.e., horror, comedy, dramatic, science fiction, etc.) the music library has the same feelings in a musical sense. The Production Music Library needs, to be of any use whatever, to have the rainbow of musical moods or the producer can be left with only half a source of production capability.

To give some examples of the different categories a good production music library should offer, see Figure 1.

Let's take a look now and see specific examples of how the recording studios can keep happy clients using a Production Music Library.

For example, a local travel agency wants to produce a radio commercial on a budget for its tours to the Carribbean. Now, instead of an originally produced score, the studio can offer its Production Music Library as the music source. A good music library, will in fact, have a variety of musical selections that can be used for these spots. You can use a contemporary selection from the library or be even more creative with a good library. The better library will have additional things, like a "Reggae" sound from Jamica, or "Steel Drums" from Trinidad. Now, if you're a recording studio in Des Moines or Milwaukee, getting a steel drum band might be a little difficult unless you have a diversified Production Music Library.

Here's another example. It's New Years and a client wants to do a spot and, of course, the music to "Auld Lang Syne." Your recording studio could become very popular very quickly if you're the only one in town that can provide the music for the ad agency.

Or, say the giant manufacturing firm downtown is doing a slide show on the new pile-driving equipment they've got out, and they need that heavy, pounding music everyone associates with those things. If you have the right music library, you're ahead of the game.

Of course, just about everyone in the communications industry, whatever they're doing, wants the contemporary feel to their productions, but don't have the contemporary funds to record these days. So, a good Music Library should have some things that are really up-to-date. Your recording studio should be able to provide them.

How about the costs involved in using a Production Music Library in your studio for your clients? Now everyone knows, or at least should know, that one of the big reasons Production Music Libraries exist in the first place is the copyright law. You, as a producer, cannot just go out, buy a movie soundtrack album or some such thing, re-record a couple of cuts off it for your clients, and expect to remain either in business very long or out of jail very long. Hence, Production Music Libraries provide you — the studio — with the ability to provide copyright protected music at nominal cost for your clients on a re-sale basis. You purchase the library from the producers on recordings of discs or tapes, and then pay the library owners "clearance fees" for use in a variety of ways. You can pay either a needle-drop basis, a per production basis based on length with unlimited use, or an annual basis with unlimited use of the library in all your productions for one fee. Of course, based on the library you acquire, the fees will vary. However, they should not be too exorbitant. One should never really have to pay more than 800 to 1,000 dollars in the worst case for annual agreements for unlimited use of a library by your organization. Typical needle-drop fees are in the \$30.00 to \$40.00 range for individual uses, and per production rates are typically based on the length of the overall production.

For example, unlimited use of a Production Music Library for a 15-minute slide show might be as much as \$125.00.

Now, what makes for a good music library as opposed to one that has limited uses for the recording studio?

Well, first of all, a Production Music Library, by definition, needs to be fairly good-sized. Because of the great diversity of music necessary for broad application, look for libraries that have at least 75 records, though more is desired. Remember, you never have to buy all records in any library, but the bigger libraries give you all the benefits of choice. Select a library that continually updates. This means that the library will be on top of musical trends and will provide their recording studio clients with the most current music at the most reasonable cost. Try to obtain a Production Music Library that records currently in 24-track. The older, typically British-based music libraries still tend to record in 8- or 4-track.

It's good advice to stay away from the smaller five- to ten-record "Production Music Libraries." These are not Music Libraries in the true sense of the word, but merely small collections of one-composer tunes. While being contemporary in nature, they are almost always limited in scope, style, and diversity of music. Avoid music libraries that do not have printed price lists for their clearance fees. Some libraries, while having printed price lists for needle-drop rates and per production fees, note that the annual agreement prices are available "on request." This simply means that they will quote you an annual unlimited use price, after they've spoken to you and find out how much money you have. You may wind up paying three, four, or more times as much as others using the same library for the same type of use in your industry. There is no reason the annual agreement figure cannot be given on the price list.

Although it is difficult, try and be sure that the people selling you a Production Music Library are actually selling you something they own or legitimately represent. Stay away from recordings that do not list composers or publishers on the album jacket or label. For foreign music libraries (usually British) represented in this country, it is a simple matter to write the source and check on the reps here.

The Production Sound Effects Library

Again, a unique source of inexpensive audio assistance, the Production Sound Effects Library is something the complete recording studio should not be without. Whether it's an Atomic Explosion or car skid and crash that you need, you do not want to go out and get them yourself if you can help it. That's where the Sound Effects Library comes in.

The professionally produced Sound Effects Library typically comes on discs and each sound effect is individually recorded, and separated from any other effect. If you want a city bus door closing, you should have just that effect and not the effect of a city bus door closing with a downtown street scene in action in the background. A good Sound Effects Library should, of course, be as complete as possible. However, due to the nature of the product, and the fact that sounds themselves don't change, sound effect additions are minimal and therefore, most Sound Effect Libraries are small in size. The largest sound effects series runs about 30 or so recordings. A good sound effects series has an accurate sound effects catalog for quick and easy reference. There are no clearance fees to be paid for the use of sound effects on such things as TV and radio spots, or even films and slide programs, so there is no worry there about costs. However, having a good sound effects series in-house gives you that chance of additional income for making tape dubs of the sounds, for individual uses for your clients. This is something you should not overlook.

There are not really many sound effects series on the market today, and you should not have trouble contacting them.

So, get a good Production Music Library and a good Sound Effects Library and when you clients want to save money in their productions, you can be there with money saving products and services to offer.□□□

Jim is one of the good ol' boys of Nashville. His engineering career stretches back some 18 years to the days of mono mixing. He's done everything from pop to R&B to disco—and, of course, country. The aviation industry gave Jim his technical background. But he's also prepared himself by playing four or five different instruments. Some of the names on the other side of the glass from him include Bob Dylan; Simon and Garfunkel; Peter, Paul and Mary; Loretta Lynn; Johnny Cash; Don Williams; Marty Robbins; Conway Twitty; Ray Price; and Roy Clark.

ON SPECIALISTS

"Let me say that I have sympathy for them, because they're missing the rest of the world of music. They're locked into one thing and I got it all. I have done four different styles of music in one day. I did a disco record that got to number six on the Billboard charts, 'Dance With You.' In the same day, I did a number one country record. You don't listen to the same kind of music all the time. And I don't want to listen to the same kind of music all the time, either."

ON OVERPRODUCTION

"'Swarm.' That's my term for overproduction. I've had producers who have turned and said, 'Well, how many tracks have we got left?' You may look at the chart and say, 'Well, we've got nine tracks left.' He'll say, 'Great.' And he looks into the window of the studio. 'Hey, let's put an electric piano on.' Not because the electric piano fits the song and has a place or meaning in the rhythm or in the feel of the song, but it's because he sees one in the room and we've got nine tracks to go. And that's overproduction, abuse of multitrack recording. And that I don't condone."

ON PLAYBACKS

"I actually mix. I don't load tape. I like to sit down at the console, set my monitor levels equal and put the band together and get a monitor mix in the control room that sounds as close as I can make it to the record, so that the producer and the artist and the musicians can hear and understand what they're doing and correct their mistakes. I'm an old mono mixer. And that's what built mono mixing."

ON TAPE

"A competitor of 3M has stated that 3M has a greater print-through than their product. It's my opinion that there is no greater print-through on the Scotch® 250. It's just not masked with modulation noise. There also was a comment that the competitor's tape was brighter, when in fact, there was just more third harmonic distortion in the 10 to 12 kc range. I am very stringent on monitoring in the control room. And when I hear a signal off the floor, I want it to come back off the tape the same way. I don't want it to be embellished with third harmonic distortion to make it brighter, or modulation noise to confuse the bass line."

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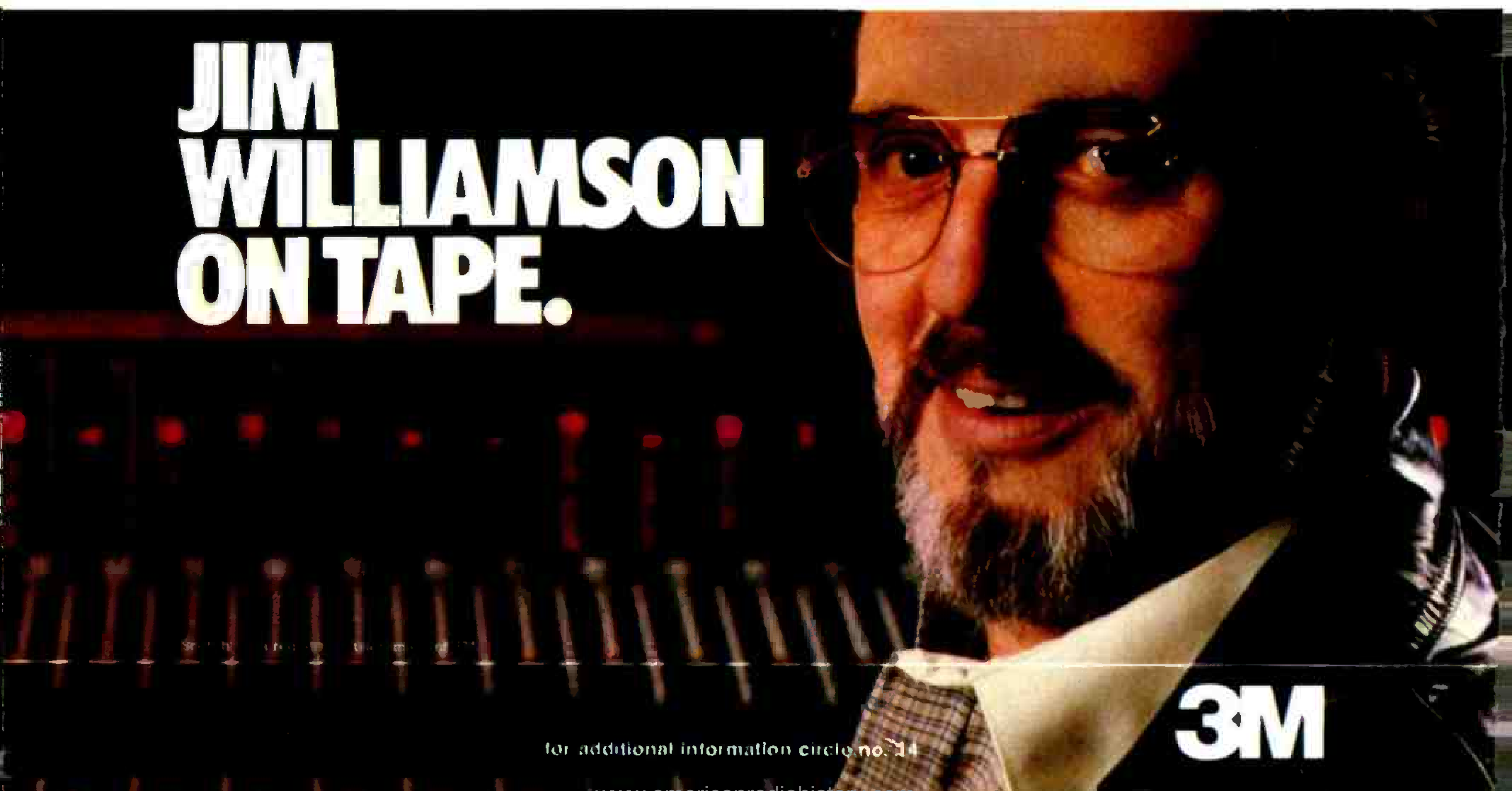

. . . at Olympic's Number 2 Studio —

George Chkiantz

by Mel Lambert

There can be few engineers currently working in Britain with the depth of experience achieved by GEORGE CHKIANTZ. Since he began his career as a tape-op/second engineer at London's famous Olympic Studios — alma mater for many leading proponents of the art of recording — George has worked free-lance with some of the most innovative bands and producers to emerge over the last decade. He was involved on early sessions with Led Zeppelin, Joe Cocker with Denny Cordell, Jimi Hendrix, Joe Boyd, Spencer Davis, and later Traffic, Family, Jimmy Miller, The Small Faces, Cat Stevens, Murray Head, and the Rolling Stones. His Gold Record achievements include one for the Stones' "It's Only Rock and Roll." But, he is equally at home with audio technology as well as the art of creative sound mixing. He spoke with us between sessions at Island Basin Street, a favorite studio and one in which George has worked on and off during the last ten years.

R-e/p (Mel Lambert): Let's begin with perhaps the hardest question of all: What has been the major influence on your career?

George Chkiantz: I think that I was lucky in that I happened to get into a studio, Olympic, which had been founded on doing jingle and film work, and which had engineers doing large-scale session musician work. Those type of sessions had to be conducted efficiently because it was money that was accounted; if you started late, you were liable (at least in theory) for the musician's overtime. Which could be expensive, very expensive. Or in the case of jingles everything is done on a minimal budget — it's a very competitive market, and they will go to whoever does the best job in the shortest space of time. As far as the client was concerned, just so long as they could do their three-minute, two-minute, 30-second, 45-second, 7-second TV ad in two hours and four different products, that's what really interested them.

And that, of course, gave me a hell of a start — being able to get a good sound and a balance in the minimum amount of time. Speed was of the essence. When the pop market in Britain started to become more-and-more important, a lot of studios didn't really have either the studio managers or the people in charge with the expertise needed to run those sort of sessions. But I had been trained at least to know what it was like to be efficient.

R-e/p (Mel Lambert): This is back in the late sixties, when Olympic was still four-track?

George Chkiantz: This was four-track, yes. In Studio One we had an Ampex AG-300 with transistor electronics, two 351s — a mono and a stereo, with a spare amp and headblock for the second machine — and a couple of ReVox G36s. The smaller studio had a valve (tube) AG-300 four-track, a valve AG-300 two-track ¼-inch machine, and a mono 351 with valve electronics. Eventually we got a four-track playback-only machine, made out of one of the old AG-300 decks. We were all Ampex at the time, apart from the copying room, which had an EMI and an old Philips deck.

The reason against European multitrack decks, I remember, was that we were attempting to build-up American clientele, because at the time the studio rates in England were very favorable to Americans, including musicians' rates. The idea was that every American producer knew Ampex, whereas a Studer was a totally foreign machine. All it would have needed was one trivial breakdown on a Studer, and it was felt that our country cousins would say: "Why the heck don't you use a sensible machine?"

George Chkiantz

On the other hand, if an Ampex broke down they just said: "Well, it's just an Ampex going down again." In fact, that happened relatively rarely.

R-e/p (Mel Lambert): So that sort of background at Olympic was beneficial when the time came for you to start engineering on group sessions, as well as orchestral work.
George Chkiantz: It gave you a great attitude, one that I haven't found in many of the English studios I've worked at. A kind of trouper-type attitude that the show goes on at all costs. Whatever happened a session did not stop; it was the ultimately important thing. It wasn't only time that is money, but that reputation is money. It was something that Keith Grant (Olympic studio manager) really felt we could do, that there was no excuse for slackness.

We were talking about the days in which other studios in England weren't as well equipped as they are today. At Pye, for example, you had to patch the mike lines into the mike amps, the mike amps into the EQ, the EQ into the channel faders, and so on. Their desk and control room equipment consisted of an awful lot of rack mounted valve electronics, which had to be linked together. All you needed was a fault in the patching and you could spend the next hour trying to find out which patch cord had gone duff.

Whereas the original desk at Olympic, which was designed for six-track film work by Keith Grant and Dick Swettenham, was just full of clevers, short cuts, and excellent goodies. It was quite a small wrap-around desk, with modules at three different angles: fairly upright EQ modules, then the echo send modules and flat faders. The desk had been arranged so that you could just about reach the knobs furthest away from you on the side wing, without leaning much from the center line.

R-e/p (Mel Lambert): The Rolling Stones were one of the first rock bands to use Olympic, and who really opened up the place to the group "scene."
George Chkiantz: The first album that the Stones did there was "Between the Buttons" in 1967, while I was still a tape-op. Glynn Johns was free-lancing at Olympic as a producer/engineer, doing most of Andrew Oldham's sessions. Which was the Stones, Small Faces, Pat Arnold, Chris Farlow; dozens of people like that. Eddy Kramer, who had joined in '66 got himself into quite a niche with Jimmy Miller and Chris Blackwell's artists, including Spencer Davis and Traffic. And Eddy worked on Chas Chandler's early sessions with Hendrix. The majority of Jimi's first album was recorded at DeLane Lea, but we did all of the second album at Olympic in 1968.

We also did a couple of later sessions for the Beatles, which resulted in "All You Need Is Love," and "Baby You're A Rich Man." However, I don't think the Beatles liked Olympic very much.

R-e/p (Mel Lambert): Perhaps the Beatles were spoiled by Abbey Road, where they recorded most of their sessions.
George Chkiantz: I think Olympic was a different kind of studio. I always thought that the fact that the Stones liked Olympic, and the Beatles didn't, was very symbolic of the kind of studio we were. The Beatles were always a more lyrical, appealing band. At Olympic we were much more infected with the Stones' disease, which was more a sort of controlled anarchy. It was rather weird when the Beatles first came in to do the "Baby You're A Rich Man" sessions with Keith Grant. They booked about half a week and he hustled them *unmercifully*. Keith had two aims: he wanted to make sure that they knew his studio was tops; but he didn't want them to use it.

R-e/p: Why?
GC: Because, at the time, he already had the Stones locking up studio time. All these bands usually insisted on using the main studio, which Keith had built for film work. It's pretty difficult to break into the film market, and he always saw the band market, mistakenly as it turned out, as a short-term phenomenon.

Between the Stones and Traffic and so on, we were running bookings maybe three or four months ahead. When the film guys couldn't even get four days together to do their orchestral sessions, Keith found himself in the position of not actually doing the sort of work he preferred, and also that he wasn't getting into the market for which he'd put in a lot of time and money. He really felt that between the Beatles and the Stones fighting for studio time, we'd have two bands that were keeping the studio fully booked for two years.

R-e/p: How did you break into session work from being a tape-op?
GC: Determining the point at which I became engineer is perhaps not quite as easy for me as for other people. Being relatively competent technically a lot of the mixing engineers would ask me for ideas to make something sound more interesting. I would try and freak things out. So I gained a lot of experience in that sort of way. I really did know how the patchbays worked, so somebody would have me patch his gear up for him . . . and then spend evenings trying to work out what I'd done.

R-e/p: Of course, this is still in the days of four-track without Dolby. How complex a mix could you build up?
GC: We could get three to four generations plus the mixing done before tape noise and general imperfections got too dreadful, because our machines were in a good state of tune. You could only take one generation of track bouncing, because the sel-sync playback was rolled-off quite viciously at about 10 kHz to avoid howl-round. However, Eddy Kramer, for example, didn't do that at all. He would tend to put down two guitars, bass, and drums, directly on four-track, and leave everything else to mixdown. Which we all thought was a waste of time. But when he stopped pre-mixing right at the beginning, Eddy began to get slightly more successful results. It wasn't so stupid really; there was always this problem of how could an engineer possibly pre-mix accurately to take account of instruments that nobody has yet thought of putting on a track.

But you couldn't help it. We'd lay down the two-track originally, overdub onto the remaining two, then we'd heave in the four-to-four transfer machine. Quite often we'd mix live on the first transfer, so that would give us a pair of phantom tracks if it was a stereo balance. And then we'd overdub on the second tape, and add more tracks while the final mix was happening. It must be remembered that, back in those days, the mono mix would take three hours per song and the stereo mix would be takes later.

Nobody worried that much about the stereo, it was the mono that sold.

R-e/p: You're often acknowledged as "rediscovering" the phasing/flanging effect, used so creatively on The Small Faces' "Ichycoo Park." How did all that come about?
GC: The phasing happened largely because of George Martin, who'd been using an ADT effect at Abbey Road. This was extremely easy to do on a Studer J-37 because the J-37 has a completely separate sync playback output. You would feed that output through the EMI BTR-2 running at 30 ips. It just so happened that the BTR-2's head gap at 30 ips was roughly the same as a Studer at 15, which has quite closely spaced heads. The net result was that you got yourself into the area where ATD, phasing, or whatever starts to happen. I think they had certain odd phasing effects that came out, mainly because the BTR-2 was pretty knackered and had a slow start time.

And they asked for this effect during "All You Need Is Love" sessions. The main difficulty was that Olympic couldn't do it because we couldn't get a simultaneous sync replay from our Ampex four-tracks. What we eventually did was to find an amplifier that

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was powerful enough to drive an Ampex capstan motor, and then hook it up to an oscillator to give us varispeed. I had to do quite a lot of work figuring out just what sort of head gap was going to be needed. Then I just had to derive a proper sync output from the original Ampex. By combining the delayed sync with the proper replay signal, hey presto: controllable phasing.

The variable speed, in fact, wasn't particularly critical, whereas I thought it would be. Line-up on the machines was far more important. In order to get the effect to work, because of all the difficulties with the re-worked record head, I used to do a special line-up on all the machines, so that we had everything coming out at the same level and, as near as I could make it, the same frequency response. An advantage was the fact that the four-track AG-300 had a heavy flywheel and therefore had quite a lot of inertia to change. The oscillator also had a very slow drive to it, so the combination of the two made for a very broad sweep that we used to get on those early tracks.

After that I did a lot of work playing around with multiple tape-loop techniques to freak out the phasing in small subtle aspects. Not directly heard as phasing but aspects of, say, changing or altering the echo, or combining it with other effects, such as a Leslie cabinet.

R-e/p: Trident was the first studio in London to go eight-track. Did Olympic follow suit pretty quickly?

GC: In fact what Trident did was to buy a 3M eight-track that ran at 12½ ips, because it was an American 60 Hz machine and the studio couldn't get the conversions for it. However, Keith Grant, of Olympic, felt that four-track was an adequate standard, and multitrack machines using any larger tape width would give problems. (Anyway we couldn't get one cheap enough, which was probably the real reason.)

I didn't find that eight-track offered much of an advantage for group work; it could be pretty awkward to set up. If you track two guitars, drums, bass, and piano, you've only got three tracks left for solos and vocals. You practically run out of space before you've started. Right-to-eight transfers weren't very easy because of the noise build-up. Sixteen-track strikes being a good all-round compromise: the track width is nice 'n' wide and causes less drop-outs; there's enough room for overdubs and fiddling around; and consoles can be kept to manageable, people-size dimensions. Nevertheless, I did a lot of good sessions on eight-track, including "Led Zeppelin II," Soft Machine, Family's second album, "Family Entertainment," and quite a lot with the Stones.

R-e/p: Did you find yourself altering your microphone technique when more tracks became available? Obviously, with sixteen-track you have more space for stereo miking on drums and other instruments. Blumlein/crossed-cardioid stereo techniques?

GC: Contrary to popular belief — especially in England — I don't believe that strictly Blumlein-pair miking is the only way to record stereo. I've heard classical records that have received rave reviews for the beauty of their coincident-pair recording techniques, and even I can tell that they've used separate microphones. I can hear that the trombones, for instance, have been shoved up a little bit late in the mix, which you certainly couldn't achieve with coincident-pair. However, the phase coherence that crossed-pair gives is very satisfying; the image is very good and doesn't seem to jar as much as mono panpotting. A lot of the trouble there, I feel, is the design of the panpots themselves — they're very crude devices and more development on what they actually do could make their effect more precise. Not only the actual 4 dB or 6 dB level drop in the center of the pot, but also phase-shift effects at various positions.

Also, the sound you obtain with Blumlein/crossed-pair miking may be very real, but I still don't find it that natural — especially on an instrument in a dead studio, because that isn't very natural anyway.

R-e/p: Isn't this getting to the heart of the matter? Should an engineer try to achieve as faithful a reproduction as possible of how an instrument actually sounds in real life? Or re-create something slightly artificial — but nevertheless pleasing — on vinyl?

GC: Certainly you try and capture the crack and power of, for example, a drum kit. But without a really high-power amplifier and monitors you aren't going to achieve it. Nor would you probably like it if you could achieve it; it would just be too big. You may feel the need to get over the effect of what a drummer is doing, but you don't necessarily want a photograph or a hologram because the average listener at home couldn't cope with it coming out of the speakers. I would suggest that what you need in order to trigger the required emotional response doesn't depend upon the analytic quality of the sound. If the *impression* works — it immediately grabs you and gets you involved with the music — then you think that you've heard the greatest recording. After all, it's done the job for you. Whereas if you heard the most analytically-correct recording, I think you'd find it was lacking in something. But, of course, the recording medium is far from perfect. If it were correct in every way, perhaps the analytical method would be better. You have to live with what the average listener has in his home; we can't all afford super hi-fi systems. What you're after is to paint a sound picture, complete with perspective, that can be contained within the medium at your disposal, but which still creates the effect a producer is after.

The most real image is the one you can form in your own mind, for which you require auditory cues. Obviously, the mix has got to sound something like the original, but to me it's more important to get the attack and the energy content that the musicians are putting out into the music. The idea is to give a listener the combinations and complexities of the blend, rather than every subtlety in the sound — which you *could* do by multi-miking every part of a drum kit, for example.

There aren't any hard and fast rules about miking. I don't think any one technique, or the use of a particular microphone, is the only way of doing something. You've got to develop and adapt to the situation; it's what you want to get out of the sound, which may well not be reality. Certainly in rock music, where you're trying to create something that could never have existed — however much a musician may try, he cannot play six guitars at the same time. So you're making the best of what you've got.

R-e/p: You left Olympic in 1973 to go free-lance. Better pickings?

GC: For many years Olympic didn't appear to mind, so long as their sessions got priority. My attitude was that if a client phones up and wants to do a session with me in such-and-such a studio, and the studio says they can't do that date, because I'm off working with the Stones, it isn't exactly a bad advertisment for the studio. Prior to going totally free-lance in the middle of '73 I worked on many outside sessions. Quite a lot of those were the Olympic clients who simply couldn't get time to work there, but wanted to take the engineer out with them. Obviously that developed into a situation where there were clients who preferred not to work Olympic but still wanted Olympic engineers.

R-e/p: That was part of a growing trend in the early Seventies toward engineers not being associated with one particular studio. A free-lance boom which is much less of a phenomena now because there are less sessions around anyway, and more staff engineers?

GC: Yes, there are more staff engineers now, and less money, I think. After all, if you pay for a studio with an engineer, you don't get much of a discount if you don't use him. Back in those days everything was that much cheaper generally; there seemed to be more money around.

I think the situation changed in a rather interesting series of ways. In the early days at Olympic we had no session clashes. Then all of a sudden this excellent lady who coordinated the session work left, and her successor just didn't have the same kind of ability. The management suddenly noticed that they didn't have engineers, or that somebody had to be refused a session because an engineer was out. So they started saying that we could continue with free-lance work, but could only give confirmation 24-hours from the start of a session. It was really largely over this that I left Olympic, because the rates of pay were certainly lower than other comparable studios, and, of course, we'd all been living quite happily on free-lance

". . . It's all very well having a desk that can do everything, but if you're into something inventive you're going to try and do something that the desk wasn't designed to do. A well thought out patch bay is essential! . . ."

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George Chkiantz

". . . one of the disadvantages with being freelance is that you can become stale and possibily typecast . . . you tend to work within a certain field . . . with people you already know. You can get very slack . . ."

earnings as well.

However, one aspect I did like was having a studio of that calibre as a home base, a place where I had a very privileged position and pretty free access to the building. I had complete freedom to do more or less what I liked when I liked, other than sessions required by Olympic which is perfectly fair. That situation as a staff engineer allowed me the time to work with top-grade equipment and to work out special effects and things like that. That's possibly one of the biggest disadvantages of being a free-lance engineer. By the time that all the gear [in a standard studio] is lined up and working, you may be talking about three or four hours to get a given special effect together. If it's going to cost the client £200 in studio time every time you open your mouth, you're not going to try that much unless you're pretty certain the effect is going to work.

When you're working out on a limb by yourself, creating new sounds, the essence of it is that you don't know what's going to come out the other end. For example, effects that maybe required two or possibly three generations to develop. Subsequent to the "Itchycoo Park" session, I was working on the idea of matrixing the stereo signal in certain ways, using effects on the matrix signal, and then re-combining it to produce a stereo which was very different from the original. But this normally required you to do an intermediate stereo balance in order for the final mixdown with effects to turn out right. It was that kind of thing that might have produced an absolutely fabulous result, once you got it right six weeks later. But you could easily spend maybe two days getting all the effects to work, and God knows how long mixing in a studio that you don't know too well.

R-e/p: That's possibly one of the major disadvantages of being free-lance. What about the positive side. Did it give you more freedom to work with bands that you liked? Do you tend then to pick and choose your sessions?

GC: On the whole anything I like I get to do, and anything I don't like the sound of just disappeared. By a strange coincidence most of the music that I wanted to do and like would appear when I had time to do it. Or people would move around to produce the time. During that period of middle to late Seventies I worked on a lot of the Family albums at Island Basing Street and quite a few King Crimson sessions; I did their last studio album "Red," and their second live album, "USA," in the States with various mobiles.

That's a story in itself. We were officially going to use a Record Plant mobile. I arrived in Milwaukee ready to go, only to discover that the mobile had blown a main bearing in Chicago. The driver had the truck towed into a garage to get it fixed. Unfortunately, the mechanic didn't fix it but locked it up and went away for the weekend. The next Crimson gig we had to record was in Toronto and the Record Plant mobile still wasn't ready. So we hired one from there which was the most extraordinary mobile I'd ever seen. I can't remember its name, but it had a very long Neve console set up along the length of the van. Which made the monitoring very interesting. The speakers were hung above the desk, pointing down at you; the subtended angle must have been 150 degrees.

Then somebody said, Oh by the way, just in case we want to do a quad mix of this could I sort out a decent track assignment. Anyway I worked out some kind of a layout for percussion with which we could decorate the back soundfield. This was all happening five minutes before the gig, and it was very difficult to do very much about it. But I laid it all out and we came away relatively happy with the balance. Then the Record Plant mobile caught up with us. Having committed myself in Toronto to a certain track layout I was somewhat stymied when I discovered that the Plant's desk didn't have panpots. Nor could you get a phantom center by pushing down two routing buttons, because it shorted out the busses. In order to maintain the same sort of track configuration I used about 40 little Shure mixers in place of panpots. I had nightmares for the rest of the tour that someone would decide that it was musically essential to edit between the tapes done on the Neve and the tapes done in the Record Plant mobile.

R-e/p: You also worked on the Continent, mainly in France?

GC: I worked at the Chateau de Herouville with Phillipe Rault, who had quite a thing about bringing back American blues and jazz artists. Phillipe was an incredibly tall, lanky character from Normandy, an absolutely lovely bloke. I recorded quite a lot of work in France, because I speak the language fluently, which helps with communication. Phillipe had a certain number of albums to produce, including a couple of Micky Baker, the guy who wrote "Love Is Strange" for the Everly Brothers; he's also done many, many guitar handbooks. We also did quite a lot of subsequent work with Micky in England. There was Memphis Slim, with whom I did two albums and Julio Finn.

R-e/p: A pleasant break from rock sessions. Do you pride yourself on being able to handle just about anything that comes along?

GC: Yes. At Olympic I'd done relatively few orchestral sessions, which in some senses I regret. But I've also recorded classical music in my time. That, of course, was what the great thing about Olympic was, because the studio made its bread and butter money from jingles, films, light orchestral music, and some classical. Most engineers, however, couldn't stand doing the classical sessions because they found them boring. I happen to like classical music so I got landed with the sessions when I was a very young engineer. Unfortunately, the studio was too dead to take large classical orchestras; the maximum size we were involved in was maybe 20-odd musicians with pretty simple balancing.

One of the other disadvantages with being free-lance is that you can become stale and possibly typecast. You tend to work only within a certain field and probably with people that you already know. You can get very slack.

R-e/p: How many productions a year have you been working on since you went free-lance?

GC: Something like four or five albums a year. I don't like to do more than six albums a year on a free-lance basis, because the amount of preparation and effort that goes into the kind of album I'm interested in doing makes for a pretty tight schedule. And you need to recover properly afterwards; clear the sessions out of your mind before moving onto something else.

R-e/p: Have you noticed any difference in the productivity or style of working between American producers and bands, as opposed to English or Continental sessions?

GC: Americans are usually more together and work more quickly than the English. That's largely because their musician rates and studio time tend to be much more expensive. In that sense I prefer working with English producers because they can usually understand that there's no point in hassling a musician who's got a severe attack of "tunitus." After all, it's not going to make it any easier to have somebody bawling down the talkback every two seconds: "Have you finished yet?" I find that American producers are sometimes prone to do that. By and large though, it's not all that common. When you get to work with the higher echelon of the industry, people who know what they're up to and know how the business works, all of them are pretty much the same everywhere.

R-e/p: What about the differences between working in city studios, which tend to be more of a hustle, and somewhere like the Chateau or The Manor — an out of town

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residential studio where you're working in the same environment for a couple of months. Do you have any preference for either sort of studio?

GC: They certainly are very, very different. I hate to say it — because working in the country is much more fun — but it's probably best to work in town. Everybody splits at the end of the session and goes back to their girl friends, wives, hotels, or whatever, and they aren't all congregated in the same place. It gives them some sort of release.

Recording is a very tense business; you're dealing with people's creativity, and it can start to go seriously wrong if a large number of people are holed up together. I don't know that the country idyll works that well; you certainly get a different sort of music, but I've seen some country sessions go horribly wrong. Actually, if I'm asked to work in the country I much prefer not to be in a studio but to use a mobile instead. I think that most studio monitoring is pretty rotten anyway, and I don't really see a mobile as being that much worse. Unfortunately, I don't think that mobiles are used in anything as a flexible manner as they ought to be.

Most of my location recordings have been with the Rolling Stones' mobile, plus The Island, Manor, and Maison Rouge mobiles. But I am quite fond of the old Stones' mobile, because I can reach everything I need, and know what it can do. Also, it hasn't been tarted up too much; it's a very functional unit.

The last time I used the RS mobile was at Knebworth Festival in August of last year, recording the two Led Zeppelin concerts. Which went very well apart from lack of a proper sound check. An interesting gig!

One of the reasons that I like the Stones' mobile, for example, is for the same reason that I preferred the old desks at Olympic. Despite the fact that their electronic performance may be nowhere near as good as the new ones, they were much smaller desks and had a lot more flexibility built into them, which some of the larger desks have lost. I'm a great fan of Helios consoles; that's partly because I was brought up on Dick Swettenham's desks. I do like the sound that they make, and Dick and Helios have always achieved a fairly good compromise between size and ergonomics.

R-e/p: Dick was one of the leaders in the wraparound design of consoles, of grouping all the functions that you would ever need to use within easy reach of a central mixing position.

GC: The problem with being a custom-builder like Helios is that very often it wasn't the engineers who liaised with designers like Dick. Usually it was left to the accountants, who didn't really take full advantage of the services that Helios and others were more than prepared to offer. What these people often failed to realize was that Dick worked by literally taking out a blank piece of paper and saying: "What do you want?" Helios could build anything that was practical and would then tell you how much it was going to cost.

An outstanding aspect of Helios consoles was their excellent flexibility in the jack bays. They weren't too complex, but had break points at the places where you actually needed them. Whereas others, for example, tend to put them everywhere; quite often where you don't need them. Dick would even build you a free-subgrouping console, in which any input module could be used as a group amp. For example, channels one to six could be for your drums and seven and eight as the drum masters. He built quite a lot of consoles of that nature, with either split format or with in-line monitoring.

R-e/p: Do you have any preference for working on a British-style split console, with separate input, output/monitoring sections, or the in-line format?

GC: Quite frankly, apart from the fact that you're forever going for the wrong monitor pot it doesn't bother me particularly. I sometimes swear at it quite cheerfully, but I can't say that it's really ever messed up a session, or slowed me down particularly. Flexibility of subgrouping via the remix/monitor bus is good, but it's something that you can achieve just as easily given a patchbay and decent access points. I think that a lot of people don't take full advantage of a patchbay. It's all very well having a desk that can do everything, but if you're into something inventive you're probably going to try and do something that a desk wasn't designed to do. A well thought-out patchbay is essential.

". . . Mixes are nowhere near as dynamic as they used to be . . . even on four-track where you really had to push the faders around to get the parts that had been submixed incorrectly to come out at all . . ."

One disadvantage with in-line desks is that they usually provide switching for normal recording, overdubs, and remix modes. However, during remix you might want a channel to go normal recording mode, so you need a defeat button on that particular input. And it goes on from there, until you end up with so many defeats per master reset that in fact the whole of the desk is far more complex. For me, it's much easier to fool yourself with a desk of that variety than it is of a traditional English design.

What I don't like about an in-line format is that it makes the channel modules very long. I loved the little old desk that we used to have at Olympic, which was bent at three angles — faders, echo send/select module, and mike pre-amp module. The desk had been mounted on a podium above floor level of the control room, which in turn was somewhat raised from the studio floor level. From the desk we had very good visibility into the studio.

But there are other things more important than the geography of a desk. I favor slider echo send controls, mainly because you can see at a glance what's being routed where. I also like echo-send cut switches. Because of the geography of an in-line desk, a channel module tends to follow the circuit flow. Which isn't really the order in which you set up a session. I'm in favor of having the routing further away than the mike gain above the EQ unit, because once you've done the routing you can forget about it. In fact, I wouldn't mind the routing somewhere totally different. Simply because it takes up quite a lot of room. Nowadays I think there's quite a case for microprocessor routing.

R-e/p: You also like to monitor recording levels on peak-program meters, rather than VUs. Why the preference?

GC: Basically in all recording you're dealing with a medium that has got a saturation level above which you shouldn't stray. It's also frequency-dependent. What you're most interested in is how close you are to that ceiling. I prefer PPMs because I'm used to the motion and know what I'm doing with them. They make you under-record quite a bit, say the VU people; certainly you don't push the tape so much with PPMs. But I cannot see the point of knowing, for example, how good a triangle would be at heating a room — after all a VU is supposed to be reading RMS heating! VUs may give you a closer indication of perceived loudness, but surely you've got ears to do that not meters. Meters are there to tell you whether you can get it on tape or not.

One of the things I really like about the newer JH-500 MCI desks is that they've got plasma displays, switchable from PPM to VU, which seems to me the best solution. And a spectrum analyzer as well, which is a lot more fun to watch than meters during long boring overdubs. I don't think anyone really knows how to use an on-board spectrum analyzer yet, simply because they haven't been around long enough. As the years go by people will be able to use them effectively. Very often a spectrum analyzer can point a clue to what's happening in the EQ and so on. If you've got a tape machine that isn't behaving itself, a spectrum analyzer might easily point a finger to what's wrong, whereas meters just won't. I'm all for gadgets like that.

R-e/p: More and more British and European studios are automating their consoles. How has your experience been with the different systems around?

GC: I've worked mainly with the Allison 65K programmer, although I've also used the Solid State Logic and MCI systems. But I got on better with the Allison at Maison Rouge. It was for an album that really needed automation. So we went 'round all the studios in London to have a look at the various systems. We visited RAK, Strawberry South in Dorking, which had an API, and others. Town House wasn't around then — this was a couple of years ago — otherwise we would definitely have gone there. According to Robin Black at Maison Rouge, the Allison 65K fitted to their Helios console worked very well and was being used all the time. Totally reliable.

I'd like to work a lot more with automation, because I don't yet think the logistics of setting up an automated mix are right. In that sense the Solid State floppy disk system has got better access than others I've used. In consoles as well, I'm all for a very basic system with good break-in and break-out access for doing exception things. You really

— continued overleaf . . .

do have to make a separation between the things that you require 5% of the time, and the things that you need the rest of the time. Important things should be on-board, and the rest can be left to outboard gear. It's the same with automation; in many ways the simpler systems are easier to understand and you're less likely to tie yourself in knots. As a result you'll get more out of them, at this stage anyway.

I think that automation has been thrown at a fairly unwilling audio industry by companies who don't understand the engineering requirements. I'm not at all sure that conventional fader automation is particularly useful. What I would like to see automated more than the faders are the panpots, cut/mute, echo select, EQ in/out, and so on.

Programmable mutes are a real advantage. The things that complicate a mix are cutting the coughs out of a vocal track, pulling out the clicks and bangs at the end of a guitar solo; that kind of thing. If you've been sitting there from midnight 'til 4 a.m. overdubbing some backing vocals and various other things on a track, it's scarcely likely you're going to go through and cut them out every time. And so you get very used to hearing them. I wouldn't be the first engineer — although I might be the first who's admitted it — that on listening to a test pressing has noticed a cough that you could swear blind wasn't there on the mix. Quite a few albums have gone out just like that, and all you can do is to hope for the best.

R-e/p: So the use of automation is somewhat overplayed?

GC: Possibly. Even up to 24-track you don't really require a box of electronics to take control of the faders, unless the track has been recorded badly. One of the interesting things about moving from 16-track to 24 is that you might split the drums a bit more, for example, but you find that the mix is much more static. I don't think that this is because you simply cannot handle it so easily. Rather because the split is like that you find that you can get a basic balance, the compromise is right, and you need to make relatively little changes.

Mixes are nowhere near as dynamic as they used to be say, even on four-track, where you really did have to push the faders around to get the parts that had been sub-mixed incorrectly to come out at all. In a curious way I find I need less help with mixing the more tracks I have available. Whereas the assumption from the computer people would be that you needed assistance, in fact you can achieve a static balance which for me is musically more valid. From a listener's point of view, it's very disturbing to have the balance thrown, to have things potted a lot. You may not immediately notice it in an album, but it can make the music very tense to listen to. I don't believe in "ruler-type" mixes, but the balance has got to be done subtlely enough for you to hear that it's not moving. I can still achieve a good mix without automation, and I've yet to find a system that really saves much time.

There's also a philosophical factor. The time taken to physically mix a tape is actually not that great. Where automation is most useful is right at the end of the mixing session: it's the build-up to that stage which will take most of your time. During that build-up an engineer and producer will be trying out various aspects of sound, and integrating them into the whole sound picture. But that is determined by the way in which their ideas develop, and not so much by their physical ability to do the job. If they leave something out, that doesn't bother them because they know it can be brought up later. The point of the exercise is to formulate the ideas and then get a coherent image of what you're trying to produce. Also, on most automation systems I've used, it's very difficult to drop-in onto a data track without producing some fairly violent crashes. I managed to get away with it on the Allison at Maison Rouge by using three or four data tracks, rather than the more conventional two.

The number of tape passes involved with systems of that nature is also disturbing, especially just to correct the balance on a short chorus. Floppy disk systems have an advantage in that respect, but they're not entirely foolproof. You may be able to patch together parts of a mix, which is very useful, but even so you can start running out of computing potential rather faster than the manufacturers would have you believe.

Automation will be far more handy when it can be used reliably throughout the session from the recording stage onwards. In that way it will bring an element of consistency into what's inevitably a bit haphazard. Quite often you'll do the basic tracks, everybody has a week off and then they want to hear the basic tracks the way you had them before. It's something I'd very much like to experiment with, to do an album using automation from the word go.

Using post-fader foldback would simplify things a lot with automation. Which is, in fact, my favorite method: I'm not a pre-fader foldback guy particularly. One of the things I learned at Olympic, which had post-fader foldback on its desks, was that if you had a good session you had no complaints from musicians listening through cans. Since they were the only ones actually listening to the mix in mono, it gave you a very good idea that nothing was actually violently wrong with the sound.

R-e/p: You've got some strong opinions on studio monitoring, in particular the virtues of valve or tube amps versus solid state. And also how they interface with certain loudspeakers.

GC: I happen to be a Tannoy man. I have little doubt that somebody will eventually design a better speaker; they haven't done it yet in my opinion. Tannoy may have certain disadvantages: they screech a bit in the mid-range. Maybe it's just that I'm used to them, but they are magic when — and *only* when — they are driven correctly. To me what seems to have happened a lot in studio monitoring is that any old amplifier with the label "hi-fi" stuck on it has been used as a monitor amp. And a Tannoy doesn't really sit happily on some of the modern combinations. They are a relatively low-powered speaker and I've always found that when driven by certain types of valve or tube amplifiers, for example, MacIntosh, Leak TL12, or Quad II, they can be totally magical. Until very recently solid state amplifiers didn't do this.

A Tannoy doesn't like working from the super highly-damped source impedance and will reflect quite a lot of energy back into the amplifier. It doesn't like being overloaded particularly; I'm not sure if it's the speakers so much as the crossover unit. Certainly putting 350W down a 35W speaker and crossover, some clipping must occur. Tom Hidley once told me that the original Westlake designs had used passive crossovers for large power-handling speakers. While he was willing to believe that "you English boys are very clever," there was no way, he concluded, that a Tannoy passive crossover will take 100W — it simply isn't big enough.

R-e/p: Isn't the solution then to bi-amp a Tannoy Gold or Red?

GC: Well, it could be, I've never yet tried it — I don't know if anyone ever has — but I suspect you'd find that the tweeters would end up flying across the room. Unfortunately, the impedance curve of a Tannoy tweeter falls with frequency until it reaches practically nothing. So just one turn-on thump and that would be it; you'd be dissipating about 6,000W for a brief, glorious instant.

R-e/p: You've also been working on what you refer to as "Richochet Effect" . . .

GC: . . . it's now called "interface intermodulation distortion" by Otala, and that's going to be the next hare everyone will chase. Over ten different effects have now been identified at the loudspeaker/amplifier interface. They aren't normally path-mechanisms by which a signal may be distorted in the purest technical sense; that it's not equivalent to what went in. We now have total harmonic distortion and intermodulation distortion which is fairly blatant, and everyone knows about them. But then Otala came up with transient intermodulation distortion (TID), and people finally recognized something was actually happening.

While I may have not known about his work in those days, from about 1968 onwards I could hear the effect happening in both valve and transistor amplifiers. At the time I could detect it in *all* transistor amplifiers and most valve amplifiers, but there were some in which it didn't happen for some reason. I found that certain amplifiers weren't able to handle an input signal without feedback, so that if the amplifier itself wasn't

*A concise discussion of **Interface Induced Distortion** appears overleaf. A more detailed article is being prepared for publication in the October issue of **R-e/p** . . .—ed*

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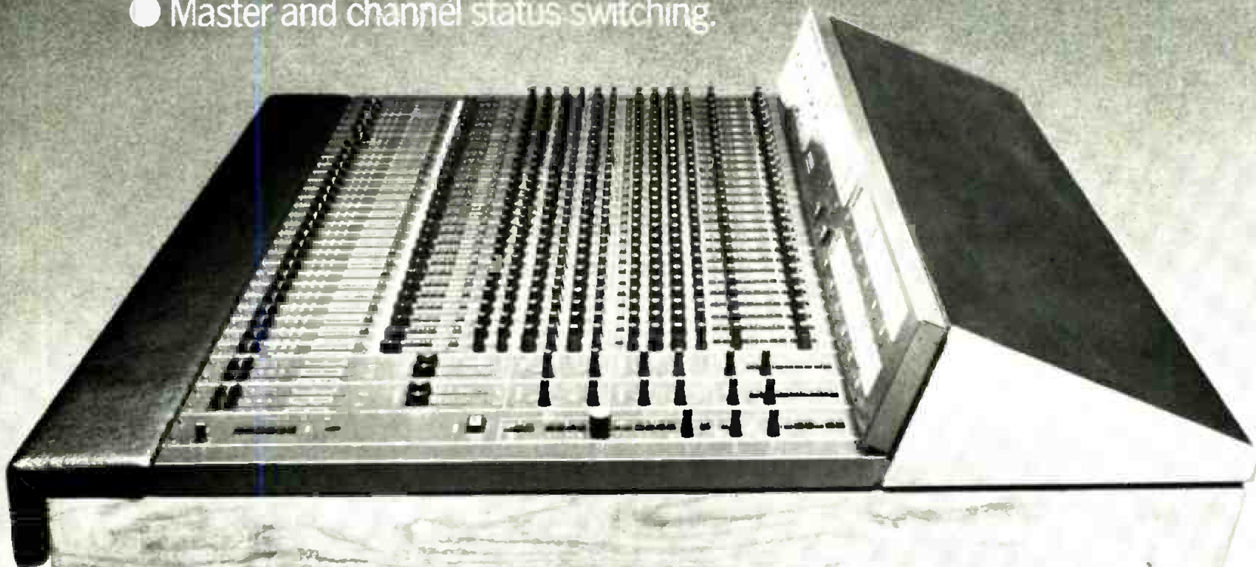

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George Chkiantz

fast enough to close the loop and catch the input signal the amplifier went into clipping. That's basically TID.

I also found that under certain conditions the same speaker feeding in one room would sound quite different elsewhere. We actually walked a monitor speaker out of one control room into another control room, and watched the character of the sound change from chalk to cheese — out of the same speaker! And the type, length, or how we coiled the cable, made no difference whatsoever.

It occurred to me that in an amplifier design using a lot of overall feedback, loading that was dependent on the room would be reflected back into the input of the amplifier via the speaker acting as a microphone. And would color the sound. Whereas if you had an amplifier in which the output was effectively disconnected from the input — i.e., no feedback — you might well do better. I think the reason I like Leak TL50 valve amps more than most is because they use less overall feedback than the majority of studio amplifiers. Being very into fine detail of sound, I'm obviously critical of the quality of electronics that a signal passes through. Far less so in the initial signal path possibly, where a particular design may or may not stop me getting a given effect, but at least I have control over it in the direct signal between the console and multitrack. However, the one place where I feel that no expense should be spared, and no compromise taken, is in that vital area past which it goes onto tape; in other words the control room monitoring.

R-e/p: Some people might say it's the Tannoys which are at fault. Have you found JBL, Altec, and UREI speakers exhibit the same effect?

GC: By and large, yes. I can change amplifiers on any of those and they show the same effect as a Tannoy. I think the Tannoy suffers most from very heavy damping, which causes it to resonate in the bass end. JBLs and Altecs survive this better and are much smoother in the bass end. An effect that worries me more about other speakers, rather the precision in the bass end, is the fact that I find the reverberant field is very critical. One easy way of testing this is to push up the echo return on a source within a mix, and to see whether the fader appears to have a broken track. In other words, you go up so far and nothing appears to happen: you breath on it and all of a sudden you're flooded in echo. If that happens I would say you've got problems in your monitoring amps.

What makes for a good balance — to my ears at least — has much more to do with the transient punch and power than sheer loudness. So if you get an artifically sharp transient produced by inadequate monitoring, you will balance it according to where that transient is in the mix. The longer you go on mixing the further down will go anything with a transient edge, *viz* the drums and vocals. How many records have you listened to lately in which it is patently obvious that the drums and vocals are a little bit back? And you can almost smell that the mix has taken quite a bit of time to do.

R-e/p: You're not a great fan of parametric EQ.

GC: Right, I often find myself in a situation where I want to do something with a sound, and my normal procedure with a parametric is to wind it up full and tweak the frequency and bandwidth to get myself in the right area. Then turn it down and see what it sounds like. Usually, I end up turning it down and turning it down and thinking nearly there, nearly there, until I suddenly feel it hit the end-stop. I've used a lot of desks with parametric EQ and none of them had a sound I liked.

So I don't use it, and go away mildly dissatisfied, which I didn't find so much with passive EQ sections. I prefer gentle slopes on an EQ — I'm not a no-EQ man — but I do think the equalization wants to be very mild and gentle. On a desk you don't want EQ that is going to vastly change the sound. Most of the time it's better to adjust or change the mike or attack the problem in some different way.

INTERFACE INDUCED DISTORTION

by David Baskind

Distortion that exists only when two devices are connected together may be defined as IID. An example of IID may be seen in analyzing a power amplifier/speaker combination. In Figure 1, amplifier 'A' with output impedance Z_O drives speaker 'S' with impedance Z_S.

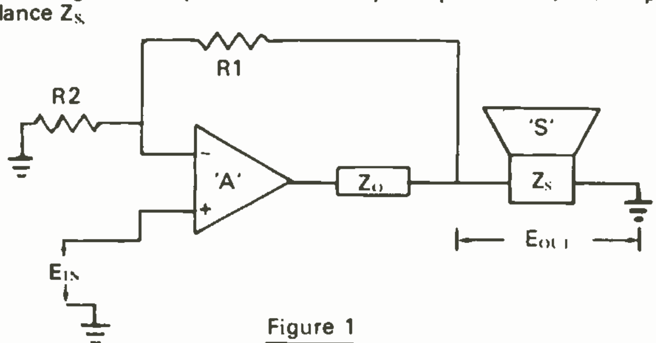
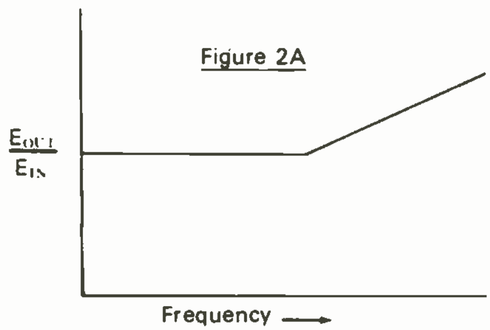
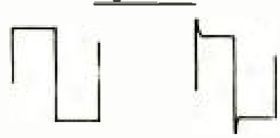


Figure 1





Input Output

Oscillogram of overshoot distortion (output) of square wave.

As in all modern power amplifiers, the gain of 'A' is normally set by the ratio of R1 to R2. (Z_O is practically zero ohms.) At high frequencies, however, Z_O typically increases above zero while Z_S may vary at different frequencies with impedance ranging from zero to both positive and negative values. The gain of the system then becomes more precisely:

$$E_{OUT}/E_{IN} = 1+R1 \times Z_O/R2 \times Z_S$$

If the capacitance of the cable connecting the amplifier and the speaker is large enough, it can become the dominant term in Z_S and Z_S will decrease with frequency.

The result of Z_O increasing while Z_S decreases will therefore be boosted high frequencies or overshoot (see Figure 2) which is one of the more common forms of transient distortion.

Most manufacturers include small inductors at the amplifier output to reduce the above mentioned effect and to prevent oscillation in the event that the speaker impedance goes to zero or negative. These inductors typically work, however, at frequencies high enough to allow the amplifier to overshoot with commonly used speaker cable. It should be noted that UREI has introduced an amplifier that greatly reduces this problem via a separate coax to correct the feedback of the amplifier. (A detailed article on Distortion, including other forms of Interface Distortion is being prepared by Mr. Baskind for publication in the October issue of R-e/p. — ed.)

Something I really would like to try one day is record an entire album using only one kind of microphone, rather than a whole variety of mikes. Possibly an AKG C-451 with a CK-1 capsule. I think it might give a "wholeness" to the sound, an evenness you miss by using all kinds of mikes.

R-e/p: Back to basics maybe. You're no great fan of Dolby either?

GC: Well, I don't like noise and the bottom end of 30 ips is a bit rough. Thirty ips is also a very expensive way of getting extra headroom and top-end performance, and you're far more likely to run off the end of a reel on a good blow. Certainly using Dolby on raucous rock and roll-type music is pretty much of a disaster all the way along the line, because it isn't doing anything 90% of the time on that kind of music. Anything above -20 dB and Dolby switches out anyway.

Splitting the frequency bands is something that you pay for; I don't think that you can bandsplit an audio signal, put it back together again and end back with quite the same thing.

Most times, however, I have to use Dolby on the multitrack but very rarely on the mix, since it's the master tape that will be going out of the building. Because of the mirror image involved in the Dolby encode/decode process, it probably manages to do a reasonable job on the multitrack. I may not like it very much, but I like the noise when I switch it even less.

dbx is something that I've used relatively rarely in England; it's still not that common. I personally love it. I find most of the pumping faults that people claim to hear with dbx are actually modulation noise. I do like the very low noise floor you can achieve with dbx — that impresses because I like having a wide dynamic range to work with. By choice I would use dbx at 15 ips, as I did on two Jade Warrior albums — "Way of the Sun," and "Kites."

R-e/p: Of course, digital recording could change all that. Apart from EMI's digital classical sessions — plus one or two others — in Britain we've had little experience of it yet. Do you like what you've heard?

GC: Even though I haven't yet worked with digital, all that I've heard may have been very impressive but still suffered from some degree of quantising noise — there's something disturbing about digital. It always tends to sound as though it's a synthesizer rather than the real thing. I suspect that the PCM code is being clocked at far too slow a frequency; I don't believe that you can cut back the passband to 22 kHz, or whatever. However, I don't find this effect on BBC FM transmissions, all of which are 13-bit PCM linked. The absence of modulation noise, wow and flutter and so on is great; absolutely superb. But there's something else happening. On the other hand the EMT-244 digital reverb is superb; great sound, easy to use, everything. So it's possible to get a good sound from digital technology. But I'm still baffled about what causes the funnies with PCM records and multitracks. Perhaps someone will find an answer pretty soon, because the future definitely lies in that direction. But only when it sounds like it did in the studio. That's the crucial test after all.

□□□

BASS TRAPS

Michael Rettinger
Consultant On Acoustics

Since the late 1970's recording studios have increasingly relied on so-called bass traps in their walls, ceilings and floors to control low-frequency absorption of sound. Broadly defined these bass traps are cavities or recesses of various dimensions, lined with one- or two-inch thick, vertically suspended, free hanging, glass fibre panels. The face of the opening is covered with a thin layer of material (grid cloth), and when installed as a pit in the floor is covered with a substantial mechanical grid capable of bearing typical floor loads.

Design Parameters

When the recess depth represents a quarter wavelength of the incident bass note (and odd multiples thereof) maximum sound absorption occurs at that frequency. The reason for this is that the air particle velocity is at a maximum at a quarter wavelength when reflected from a hard surface. At the quarter wavelength position the air molecules move at such a rapid rate, in and out of the horizontal layer, that a considerable amount of the acoustic energy of the wave is converted into heat within the frictional interstices of the material. Of course, other notes, too, are thus absorbed as their waves pass between the spaced glass fibre panels, striking the soft bottom layer and returning again through the highly absorbent environment.

The table below gives the quarter wavelength of various bass notes, and thus the required trap depth for this frequency of "suppression:"

	— Depth —	
f	**Meters**	**Feet**
30	2.86	11.38
40	2.14	7.04
50	1.72	5.64
60	1.43	4.69
70	1.23	4.03
80	1.07	3.52
90	.96	3.13
100	.86	2.82
120	.72	2.35

When a *band* of bass notes is to be absorbed, the bottom of the pit should be slanted. As an example, when the octave between 40 and 80 Hertz is to be expunged from the incident signal, one end of the cavity should have a depth of 7 feet and the other end should have one of 3.5 feet. For such wide-band absorption the cross-sectional area of the trap should be larger than when only one sinusoid is to be effectively absorbed.

The very high absorption capability of such a glass-fibre-lined floor, wall or ceiling chamber, as far as the whole room is concerned, is due to the fact that precisely where the air-particle velocity of the reflected wave is at a maximum, the sound pressure is at a minimum, being 90 degrees out of phase with the particle velocity. The reason for this is that at the bottom of the pit the air particles cannot move at all, they are facing an immobile barrier; while at the same time a doubling of the pressure amplitude results. The face of the trap, then, represents a vacuum, drawing "sound rays" into the cave, which otherwise would have travelled elsewhere, as diagramatically illustrated in Figure 1.

Bass Trap Purpose

The purpose of a bass trap is twofold — to absorb floor reflections for the bass notes of the drums and violas and other instruments with predominantly low-frequency output (when the instrument is positioned over the trap), and to lower the music level of these bass notes as they arrive in the vicinity of adjacent instruments. The primary use, then, is to prevent bass notes from leaking into microphones where they are unwanted, in multitrack recording situations. Another purpose is to provide greater acoustic comfort for the other players in the studio. Drum notes, for instance, can reach a sound pressure level of 100 dB at a distance of one meter or three feet, and violas, too, can generate acoustically uncomfortable levels, particularly for nearby violinists. An incidental purpose of a trap is to absorb really low notes generated anywhere in the studio.

Where bass frequencies are not controlled by traps there is often a need to segregate certain musicians. Drummers are placed in drum cages (Figure 2) and pianos in similar enclosures, away from the band, often to the discomfort of these performers.

From the experiences of many recording engineers, the use of "flats" or "gobos" between instrument sections to produce sufficiently clean tracks for multitrack recording has been unsatisfactory. This is often due to the ability of the sound to creep

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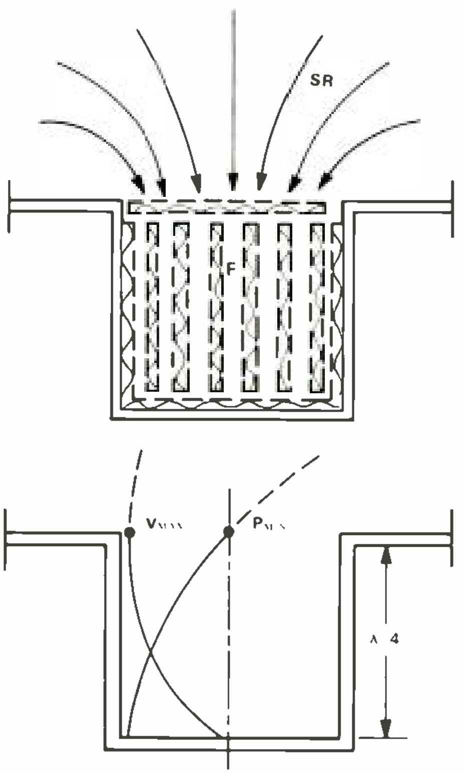

Figure 1: **SOUND TRAP**
SR: Sound Rays. F: Glass Fibre Panels. V: Velocity Variation of Air Particle. P: Pressure Wave.

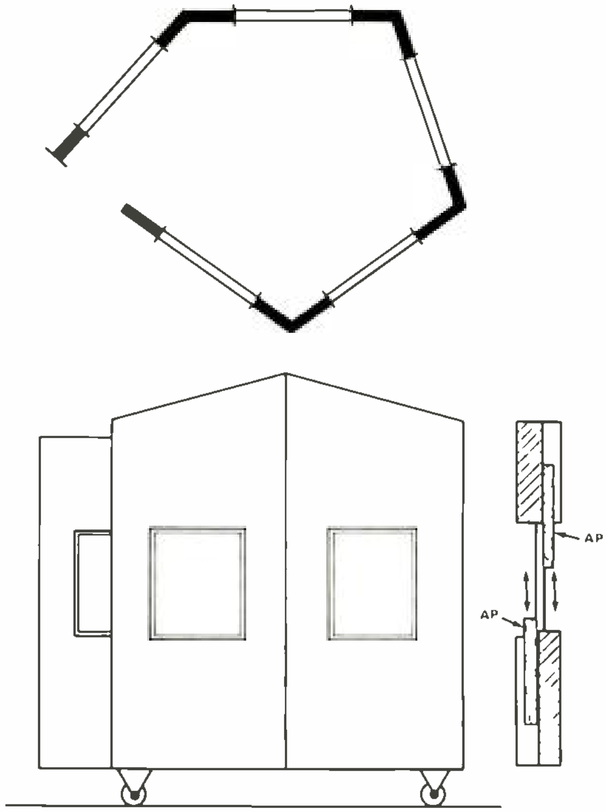

Figure 2 **DRUM CAGE ON WHEELS**
AP: Moveable Acoustic Pnaels Over Windows.

and leak around and under poorly, inexpertly or cheaply designed partitions, many of which are built off the floor, on wheels, and are not thick or high enough. Also, these devices do not allow the performer to see others in adjacent areas for allowing better timing because space dividers generally do not have windows. As additional arguments against this kind of leakage control, some engineers consider flats too cumbersome to adjust, and when not in use they take up valuable floor space unless storage was planned for in the design stage of the studio. Finally, as hand-crafted items, such units are expensive, and sometimes obscure the view of the mixer intent on watching the performers.

In this connection, the writer would like to cite an experience he had in 1943 when Columbia Pictures, in Hollywood, decided to make the "Al Jolson Story," with Larry Parks. Al Jolson had been commissioned to personally sing the important songs in "The Jazz Singer." He insisted on rendering his vocals from within the orchestra, because as a stage performer he had always been surrounded by musicians. To accommodate him a glass booth was constructed with a heavily absorbent floor and ceiling, earphones were offered to him so that he could sing in synchronism with the instrumental music. He would not wear them. Surprisingly, the very little sound which filtered through the glass was enough for him to stay in time, although it could not be heard on his track.

— continued overleaf . . .

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BASS TRAPS

Michael Rettinger
Consultant On Acoustics

Disadvantages Of Bass Traps

There are two disadvantages connected with the use of bass traps. One lies in the music sounding somewhat less good, that is somewhat less lively, rich, and full. The second relates to the performer positioned over a bass trap feeling somewhat uncomfortable, as if he were suspended in mid-air, without any reinforcing sounds about him.

The physics of sound will enable a more detailed look at these effects. In 1931, G. W. Steward, Professor of Physics at the State University of Iowa, published the "Acoustic Uncertainty Principle."* It is similar to Heisenberg's uncertainty principle in quantum mechanics. By the latter theorum one will never be able to see an electron under a microscope, because the impact of the light ray upon the sub-atomic particle moves it so much that the reflected light ray in the eyepiece can show up as only a blur. By the acoustic uncertainty principle we face a similar inability to recognize the components of a complex sound wave. Stated mathematically, both the Heisenberg and the Steward uncertainty principle may be expressed as follows:

$$\Delta f \times \Delta t = 1$$

where f refers to the frequency in the acoustic case, and to the total energy of the moving particle in the case of quantum mechanics; where t represents the time in both cases, while delta is the class interval.

As an example of the acoustic principle of uncertainty, consider a musical note. One does not hear a single frequency tone but a pulse which by Fourier analysis is found to consist of a number of sinusoids. There is a frequency f1 of maximum intensity and a range of higher and lower frequencies with different intensities. When one tries, aurally, to identify f1 there is difficulty in recognizing it because of the presence of other components. As an example, assume we wish to determine a 1,000 Hz f1 with 1 per cent accuracy. This means that one has to evaluate f1 within 10 Hertz. By the acoustic principle of uncertainty, the required time interval is .1 seconds, because 20 x .1 = 1. When the frequency interval is to be only 3.3 Hertz, the required time interval becomes .3 seconds, because 3.3 x .3 = 1. In other words, the accuracy of the frequency definition increases with time. But there is no way in an anechoic environment to increase this time, because there are no "fusing" reflections. However, by placing a reflective surface near an instrument, this time interval is increased by the minutely delayed reflection; besides, the reflection provides a greater tone intensity at the ear of the performer. Musicians are very conscious of this effect and call it "attack" or "tonal force." This is also one of the reasons why we like to sing in the shower, although in this case we have also the effect of a longer reverberation time.

What about acoustic comb filter effects when we have a reflecting surface near an instrument? Too, what about the sounding board of a piano which reflects the action of the strings? Or better yet, what about the twenty violins in a symphony orchestra? At the microphone their signals are all out of phase with each other, and yet their rendition is better than that of a single violin because of these phase relationships. Anecdotally, at the beginning of sound-on-film recording a producer thought he could save money by having the recordist turn up the amplifier level 13 dB in imitation of the signal strength of 20 violins compared to that of one violin. The effect was very discouraging when the photographed scene showed a violin ensemble where the distant performers were not musicians but mimicking extras. So, we cannot ignore the facts which have made music rich in appeal, even when acoustic spectrometry appears to show an anomaly. We have a great deal to learn about the ear and, especially, about the human brain which analyzes what we hear.

A theorist might object to such bass traps for reasons other than those already mentioned. He might say that a studio with such a trap is no longer a room, a three-dimensional enclosure, because the sound absorbent pit represents a discontinuity of the bounded space, where sound can enter but not return. By the G. Millington equation, he might say, it is not even possible to calculate the reverberation time of the studio. The reasoning would be that the equation contains the term:

$$S \log (1 - a)$$

where S is the cross-sectional area of the trap and a is its (unity) absorption coefficient, which term would become minus infinity. Actually, by our method of absorption measurements, a would turn out to be larger than unity, for the reason that some of the normally diffused signal would be "sucked" into the hole, which sound rays otherwise would travel in straight lines about the enclosure. Lack of uniform distribution of the sound energy would result in a non-exponential decay of the signal on sudden stoppage. These are the fine points, which in studios with reverberation periods of .2 to .4 seconds would probably not be noticed by the majority of performers. They are presented here for the sake of analysis of the subject of bass traps. □ □ □

*Steward, G. W., "*Problems Suggested By An Uncertainty Principle In Acoustics*," Jl. Acoustical Society of America, Vol. 2, No. 3, January 1931, page 325.

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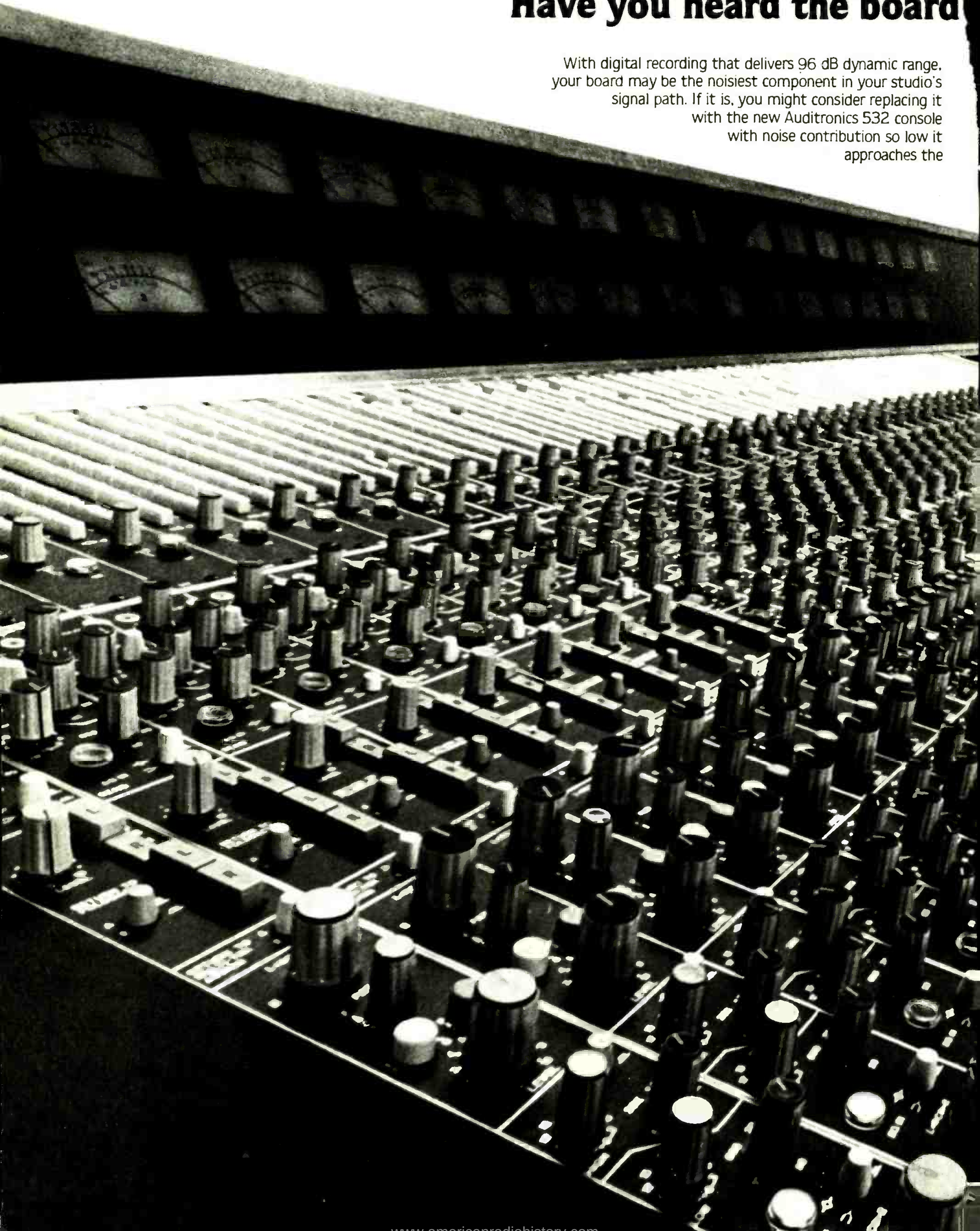


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FILM SOUND

. . . techniques for preserving SOUND PERSPECTIVE in film production recordings

by Steve Barnett

In the April, 1980 issue of ***R-e/p***, Jim Webb discussed pioneering techniques in multitrack recording of motion picture production sound. These involved primarily, a number of radio microphones placed on the actors, as opposed to the more common approach of a single mike on a boom or fish pole covering all the action in a scene. In this article, Webb and boom man Chris McLaughlin explore this latter technique, known as perspective miking, as well as methods derivative of both perspective and RF/multitrack recording of film sound.

In recent years, sound has become a more important element in the primarily visual medium of motion pictures. Films such as "Star Wars" and "Apocalypse Now" relied heavily upon their audio to convey the action as well as the mood, but epics of this scale are not the only pictures to use sound as an integral part of the overall motion picture experience. Production mixer Jim Webb and boom man Chris McLaughlin have, through the use of creative miking techniques, been able to convey style and mood in the production sound for the motion pictures on which they work. (Production sound is the audio recorded during filming.)

The framework within which they are currently accomplishing this task is that of "perspective miking." The camera is the eye of the audience. They see the action from its vantage point. Generally then, if the person speaking is some distance from the camera, he should logically sound farther away than someone in the foreground. In keeping this perspective, the environmental sounds around the action become quite important in establishing the audio viewpoint as well as the audio style and mood. The relationship of the level of the person's voice to these background sounds gives the audience a distance reference, and perhaps more importantly, helps establish the mood for the piece so that what the audience sees is also what it hears.

"The perspective is the mood," said Webb in his previous article, "whether it be an echoing hallway or a noisy street. It is the thing that gives life to the track. Without it, the movie no longer sounds like it looks."

Consequently, how the background sounds are mixed into the production track is a matter of some concern and a major part of the boom man's and mixer's art. Just as there are exceptions to the style described above, there are also exceptions to how that style is achieved, but usually it is accomplished with a single microphone with its level adjusted by the mixer and, as importantly, with its physical positioning manipulated by the boom man for the desired effect.

This style is in contrast to the multitrack format that Webb and McLaughlin developed with director Robert Altman, which incorporated the use of radio microphones on most of the actors in crowded scenes allowing them the freedom to improvise without worrying about their position in relationship to the mike. Altman used this technique to realize what many feel is a more spontaneous style of film making, however, the electret microphones planted on the actors pick up little more than the voice of the subject. They lose the environmental sounds of the background which characteristically impart the mood. Also, once planted, these mikes cannot be manipulated for effect. So with the logistics of this multitrack/RF format accomplished, Webb and McLaughlin found themselves with little room to add artistically to a picture. Though there are ways to add perspective to the multitrack format, the two felt their creativity could be put to better use on films with techniques favoring the perspective style with the emphasis on a single, primary microphone. An example of this sort of picture is "Straight Time," which starred Dustin Hoffman.

— continued overleaf . . .

Cherokee Studios, Hollywood, California

FILM SOUND

. . . techniques for preserving SOUND PERSPECTIVE in film production recordings

to that, whether it's a realistic sound, a close sound, a warm sound, or whatever. Chris has a tremendous amount of control with that mike, and he can shade the sound by playing the mike in certain ways."

Though the two will often discuss with the director of a film the type of sound he wants and the feeling it should communicate, with "Straight Time," it was the setting that set the style.

"We shot in practical locations in downtown Los Angeles," says McLaughlin, "and the environmental sounds, like the area, were depressing. Now this was the story of a guy fresh out of prison on parole, and about how tough it is for an ex-con to make it. He lived and existed in a crummy apartment in this trashy area of L.A., so we didn't want to have him living in this hole and have it sound like the best stereo hi-fi. Without sounding bad, we wanted it to have a hard edge, a gritty, almost documentary feel, so I miked it slightly off. I never gave the actors the microphone directly or pointed it right at them, so the voices were never right into the mike, and the feeling was airy. We heard the noises of the building and the street along with the voices, and hopefully we lent to the depression of the story through the sound."

The 815 *

To accomplish this task, McLaughlin used what is his favorite microphone, the Sennheiser 815, which has a very narrow cardioid or club-shaped pattern, a long reach, and, according to McLaughlin, an uncanny accuracy.

"The 815 is a perfect instrument," says McLaughlin. "It will do exactly what you tell it to do. On one occasion at a major studio, I tried to get that mike and they wouldn't give it to me. They were afraid I was going to miss because it was so directional. For that reason, you've got to be good to get the sound you want and to keep it consistent throughout an entire picture."

As an example of the 815's accuracy, McLaughlin cites a story from his work on the film, "Remember My Name," which he did with another mixer.

"He wanted me to use the Sennheiser 415, and although I've used that mike when the situation warrants, I still prefer the 815. We settled the argument in a scene where Geraldine Chaplin drives up in a car, gets out, walks toward the camera, stops about head-to-toe in the frame, lights a cigarette with a stick match, and flicks the match away. On the first take, I used the 415 and got the car door slam, the match strike, the traffic noise, and that's about it. On the second take, I used the 815 and we got the door, her footsteps, the match strike, the flare when it exploded, her drag on the cigarette, the crackling of the tobacco as it lit, and when she flicked the match, I followed it all the way down and got the 'plink' when it hit the pavement. It wasn't anything special as an effects shot, but it sounded wonderful."

Other Styles For Other Moods

McLaughlin and Webb use the 815 to achieve other styles aside from that on "Straight Time." For their work in "Movie, Movie," a parody of the films of the 1930s, they miked the actors tight and right on target to achieve the old style, full mellow sound that was the hallmark of the pictures being lampooned. In another film, they mixed their techniques to establish a shift in mood.

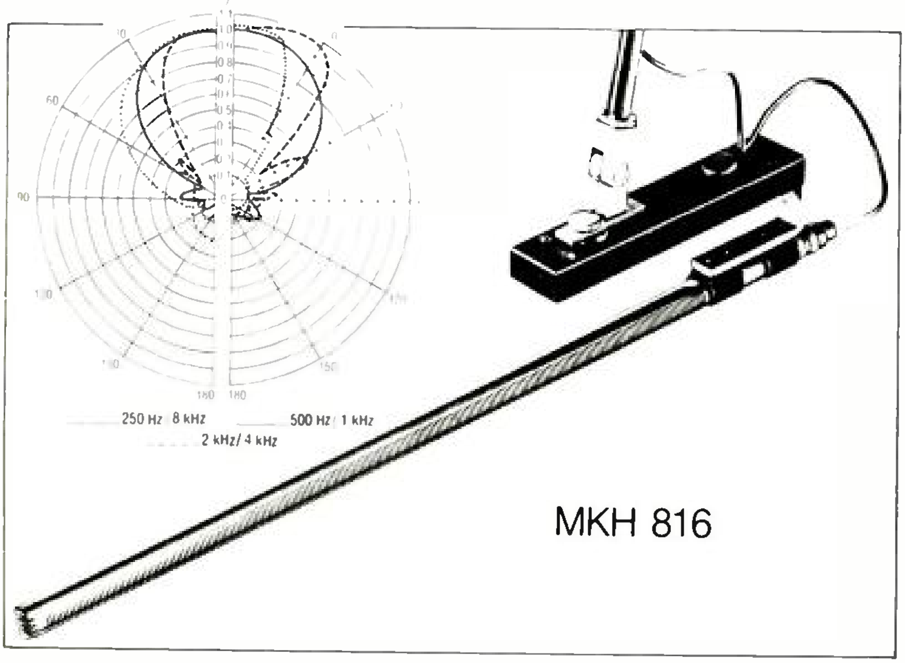
**The Sennheiser 815 and 415 mentioned in this article have subsequently been improved and have been reintroduced as the Model MKH 816 and MKH 416 respectively.*

— ***Chris McLaughlin . . . fishing***

The picture was "A Small Circle of Friends," and dealt with several college students and the upheavals on the nation's campuses during the late 1960s.

"Most of the stuff we did on that shoot was virtually dead on," says McLaughlin, but he adds that "we had a chance to do different things in different areas of the film because it dealt with so many facets of the sixties. We were able to play in different scenes where we thought we could bring something to the mood. In one sequence, Brad Davis's character is taken blindfolded through a series of connections to meet a friend who has become a complete revolutionary. They meet in the kitchen of this house in the country where this Weatherman-like group is making bombs to blow up banks and such. I miked that one a little airy to lend an anticipatory quality to the atmosphere with all the clocks ticking and the bomb making activity going on in the background. This was in contrast to the tighter sound in the rest of the picture."

McLaughlin notes as well, that off-miking on a sound stage can be less effective as the environment is so dead that the dialogue is not reflected back into the microphone. Nonetheless, we see that the mood is set aurally by the quality of the dialogue and its relationship to the environmental sounds. These sounds, however, can become unwanted noise if they are too prominent in the track. This is one reason that McLaughlin must not only work the actors' voices, but his placement as well. There is a signal-to-noise factor in film sound recording that refers not only to the equipment performance, but also to the sounds desired on the track in relationship to the unwanted, extraneous noises often found on location shoots and even in the studio. This signal-to-noise is affected by the boom man's angle of attack with the microphone. Obviously, he does not



VALLEY PEOPLE NEWS

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Vol. I, No. 1 **September, 1980**

Valley/Allison Merger Official—Paul C. Buff, Inc. Formed

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FILM SOUND ... techniques for preserving SOUND PERSPECTIVE in film production recordings

point it against any traffic background or any noise field of that nature, but he must also take care in totally enclosed areas.

The Dead Spots

"You find the dead spots," says McLaughlin. "With a Panaflex camera running in a pretty dead room, you can hear the motor, and I *fish* with the 815 and find the spot where the sound is the least prominent, and that's the angle I gun for on that take."

"Most of the camera noise," adds Webb, "comes out of the lense opening of the blimp, hits a wall or whatever, and comes back into the mike, so you've got to be able to find the null where, for whatever the acoustic reason, the noise is less present, and you've got to be able to find that hole in the few seconds between the camera rolling and the start of the dialogue. An example of this is the close-up of Robert Redford's hand in 'All The President's Men' as he made little pencil scratches on his note pad. In that kind of a quiet scene, with the camera that close, the motor noise is tremendous, so Chris had to come in from the side, and in the first few seconds, find that null where the noise was the least. An acoustic hole is there nine times out of ten, and you use it to avoid not only camera noise, but any kind of unwanted background noise."

"In 'The Long Riders,'" adds McLaughlin, "we shot a scene near Plains, Georgia, peanut country, and they have these huge peanut dryers there that whine like a prop jet. It took a little while, but I found a space where that noise was at a minimum."

"You try to get it down to where you can manage it," continues Webb, "and hope that they'll be able to get rid of the rest of it in the re-recording mix. One situation where the signal-to-noise ratio can be especially tricky is at the beach, because you've got the sound of the surf on one side and usually the sounds of the highway on the other. You have to tread a very fine line between the two noise problems, and what you need is rejection."

McLaughlin used the 815 for beach scenes shot for "The Rose," and because of its highly directional properties, was able to find the right angle to eliminate the unwanted noise.

"If you have two people sitting on the beach talking," continues Webb, "another way is to dig a hole in the sand and use it as a sound chamber for the mike. Sand is very dense material, and we've used a Neumann U87 on a bi-directional pattern set in a hole between two actors and gotten terrific results."

"Sand is a great gimmick," adds McLaughlin. "You can photograph sand, and you can't tell where the mounds are, and there are mounds of sand on the beach anyway. The real gift of sand to the mike man is that you can use it to hide your boom shadows. Once you find your spot, you build up mounds of sand to break up the shadow and you're set."

"In picking that spot," says Webb, "you've really got to watch the backgrounds, because you'll get killed by the ocean or the highway if you don't, and the sound won't be consistent from cut-to-cut when the picture is edited together."

Aside from the signal-to-noise problem, these backgrounds must be carefully monitored to see how they affect the ambience of the soundtrack and hence the perspective.

"You may be in a situation," explains McLaughlin, "where one actor is standing against a wall and a second actor is sitting with a window behind him. When the editor makes his cuts back and forth from close-up to close-up, you don't want the soundtrack to go from nice and quiet against the wall to suddenly cars and trucks outside the window, so you find a happy medium in the master shot and use the same mike angle in the close-ups."

Consistency

This brings up a major consideration in production sound recording in motion pictures. The sound must be consistent. Both McLaughlin and Webb agree that the sound should not draw attention to itself, for it then becomes a detracting factor. Changes in ambience from shot-to-shot within a scene are distracting and can subliminally throw the audience out of the suspension of disbelief so necessary to the enjoyment of the motion picture. For this reason, McLaughlin will establish his position in the master with regard to dialogue, background, environmental sound, and style, and keep essentially the same angle during the recording of the rest of the shots in the scene. In a scene in "The Long Riders," the voices of the actors themselves dictated the initial mike position.

"In the scene," explains McLaughlin, "Stacy Keach as Jesse James is proposing to his girl, and because he's got a really solid voice, I had to lay off a little bit of Stacy and favor the girl with the microphone. Then, when they turned the camera around to shoot Stacy's close-up, I still had to point the microphone away from him, but not at the camera. So I had to find a space not towards the camera, but not toward Stacy either, that would give me the same effect as when we shot the master. I don't believe in shoving the mike in tight on close-ups just because you can."

Also, as "The Long Riders" was a period piece, no modern sounds could be permitted in the track.

Another example of keeping consistency throughout a scene can be found in "All The President's Men," when Dustin Hoffman as Carl Bernstein is pressing Jane Alexander's character for information on the Watergate slush fund. The camera and lighting set-up dictated that the master be recorded with the Sennheiser 415 from below. McLaughlin held the mike on his fish pole, just off the floor, and he could not bump the floor without ruining the track, and he couldn't bring the mike up without ruining the shot. Because of these constraints the 815 proved to be too long for the task, and when it came time to do the close-ups, McLaughlin continued to mike the actors from underneath with the 415 rather than risk the change of ambience and quality that would have occurred if he had switched

— some miking jobs are harder than others . . .

to the 815 and an angle easier for him to maintain. Because it is necessay to maintain that consistency based on the recording of the master, it is important to establish correct angle as soon as possible. If the first shot is recorded from a less than desirable position, for whatever reason, you are locked into that angle for the remainder of the scene so that the tracks match from cut-to-cut. Watching the rehearsals helps to this end.

In preparation for a scene, says McLaughlin, "I'll always talk to the camera operator to find out what his framing is, what the lens size is, how much head room I have, and how close I can get in from other angles without getting in the shot. Then I look for my spot."

Crew Coorperation

To enable himself to get that perfect position, McLaughlin must rely upon the cooperation of the rest of the crew. To facilitate this, he makes it a point of knowing the jobs of everyone on the set. With a thorough understanding of their needs, he is in a better position to enlist their aid should it become necessary to have a light moved, or to ask a dolly grip to re-route his path in order to afford a better mike position.

"Part of the approach is doing favors," says McLaughlin. "I'll be on top of it if they need some help with the dolly trucks or with the laying of cable or with anything that needs doing so that when the time comes, I can get the inch I need."

"That really is a legitimate mike technique," adds Webb, "because you're not working in a recording studio where everything is locked down tight. You're in a fluid sound field where everything constantly shifts and changes, and you have to continuously make minute adjustments to achieve and keep the mood you're after. So you've got to have a super rapport with everybody in order to get the sound. When you need that little fine adjustment, it's got to be fast and it's got to be subtle, because by that stage it's just before shooting, and you don"t have a lot of time. The other half of this coin is that you have to make them aware of your needs so that they understand what you're trying to do. It's all important."

Relationships with the actors and the director are also a concern of the sound man, for if an actor is not getting picked up, a slight adjustment in his voice can make all the difference. This, however, can be a sticky situation, as there are a number of directors who prefer all the information going to the actors to come through them. It is a legitimate position, and an arrangement that must be made on an individual basis.

The Re-Recording Mix

Another reason for pursuing just the right method to achieve the desired mood during the production recording is so that the style will come across after the mix.

"We try to be consistent within the chosen style," says Webb, "so that hopefully all the sound editors and the re-recording mixers have to do is polish it up, bringing that style through the mix."

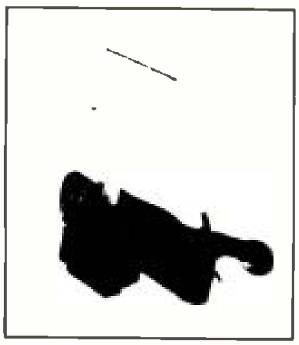
"I try to lock them into our style of sound," says McLaughlin, "the mood we set out to establish while the film was being shot. I want that idea to be the audio style of the picture."

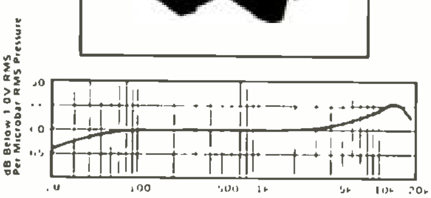
Presence tracks and effects cut-in during post-production would then be mixed with the Webb and McLaughlin concept in mind.

Radio Mikes

Even in a picture where the two are using a single mike perspective approach to recording, situations can arise that call for radio (wireless) microphones. These usually involve a shot that will not accommodate the fishpole or even the standing boom with its height and 16-foot reach. In these instances, Webb and McLaughlin rely upon Artech radio links and lavaliere electret microphones manufactured by Coherent Communications, of Los Angeles. These "Mini-Mikes" were designed by Ivan Kruglak to match the frequency response of high quality condenser microphones, such as the

Coherent Communications' Lavaliere 'Mini-Mike'





Frequency Response based on Open Circuit Sensitivity (1.3Vdc Supply)

Sennheiser 815, and their rising high end. Using these as opposed to other small electret mikes makes it easier for Webb to match the sound of the 815 for a more balanced soundtrack throughout the picture. Nearly all lavaliere mikes sound somewhat tight and dry, however, having little ability to pick up the ambience of the set that lends so much to the mood, so Webb will often open up another mike in a technique called "backmiking."

"It's an attempt to put a little air into the track," explains Webb, "to get back to the sound of the room. I'll mix the two mikes onto the same track, but you've got to be really careful with it to keep it consistent from cut-to-cut. It gets super critical. The mike used can be either the 815 or the 415, depending upon the environment."

If the signal-to-noise problem is so great, however, that the RF mikes themselves will pick up the background sounds, then no other mike is opened up and the problem is self-solving.

Another reason for the use of a radio mike is an actor whose voice is so soft that he cannot be recorded any other way. This presents some difficulties.

"Some actors, like Robert Dinero, work very soft and close," explains Webb, and need closer miking, but in a voice without a lot of timber and dynamics, the lavalieres sound bad. Without that chest resonance, they have a hard time cutting it, and so the reason you have to use an electret is also the reason that they don't work so well."

"We've solved a lot of problems," offers McLaughlin, "since Jim acquired the stereo Nagra. We go it both ways. If it seems implausible to get the sound with the 815, I'll wire up the actors, and on one track we'll put the Sennheiser, and on the other track, we'll record the radios."

"A perfect example of that is in 'The Rose,'" adds Webb, "when she and her driver are coming up to the hotel in the morning. Now if we had played this track totally perspective, with just the 815, all we would have heard was the car coming around the corner and their voices when they got out. So because we weren't sure how it was going to go, we put radios on both Bette Midler and Steve Forrest, and put those on one track with the 815 handled by Chris on the other track to keep the true perspective. As it turned out, they decided to sing inside the car as they came around the corner to the curb. When they got out, they did their dialogue and went in through the revolving door. Later, in the mix, they used it both ways. They kept the perspective with the 815, but they also kept the radio mike track of them singing as they drove up. In fact, they kept it all the way through the revolving door into the hotel, so it worked with both."

McLaughlin also points out that with radio mikes, an early involvement with the wardrobe selection is helpful, for the type of fabrics used will affect the mike performance.

"If you pick up a tie," adds Webb, "and you rub it in your hands and can hear the sound two feet away, you can imagine what the mike will hear. If you can get wardrobe to use soft materials such as cotton, it really helps."

Additional Microphones

Aside from the Sennheisers and the "Mini-Mikes," Webb and McLaughlin travel with a number of other microphones which are rarely used because of the versatility of the 815, but occasionally they are needed, and the team wants them there when the occasion arises. In circumstances where an extra mike aside from the 815 is called for, Webb will use the EQ on his board to get it to sound as close to the Sennheiser as possible.

"We hadn't used the Electro-Voice 667 in two years," says McLaughlin, "and suddenly here we are with this stage coach on 'The Long Riders,' and we both knew that it was the perfect mike to hang in the harness traces to get the sounds of that rig."

"On that picture," added Webb, "we went all stops out for effects, and we had some good effects cutters to back us up. We used the second track of the stereo Nagra and recorded the effects in sync with the action, as with the E-V on the stagecoach and in another scene on the train."

Webb and McLaughlin will occasionally use a second boom mike, but prefer to avoid that situation for the lack of control it presents in keeping the style consistent.

"Sometimes, though," says Webb, "we'll plant one of the "Mini-Mikes" on the set, hard lined into the board if we're having trouble getting a line or two being said in a strange direction. In 'The Long Riders,' we had a very wide shot where a woman came down the stairs, and we taped one of these electrets to the top of the post at the bottom of the handrail."

"It really was a great match," continues McLaughlin, "because as she came into the 815 from the lavaliere, Jim swung from one to the other, and I had found just the right position with the Sennheiser so that it all sounded the same."

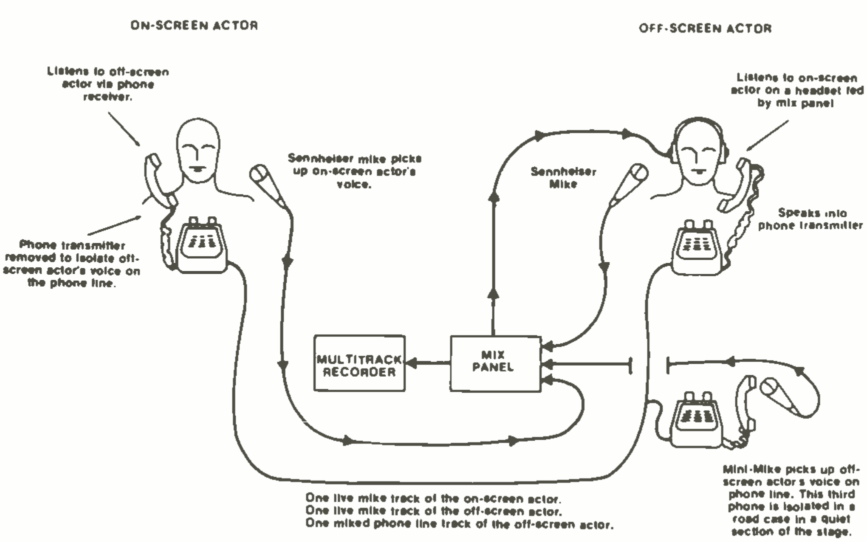
Multitrack Perspective

Another picture that relyed heavily upon more than one microphone for both effects and dialogue was "All The President's Men." Webb employed his multitrack recorder on this film for reasons that will become apparent.

"We had all the phone lines live," explains Webb, "and we took the television set audio in the scene live at the same time that Chris was working the set with the 815. We routed the TVs direct so as not to clutter up the acoustic environment."

"A lot of the picture was shot with what

UNIQUE TELEPHONE HOOK-UP USED FOR "ALL THE PRESIDENT'S MEN"



they call a diopter," adds McLaughlin, "which allows the foreground and the background to both be in sharp focus. We'd have a scene with Redford on the phone in the foreground and 19 people crowded around a TV set in the background, and we had to mike both of these areas. For the background, we used a Neumann U87 standing straight up on a stand hidden by the actors or a desk. We set it for whatever pattern was appropriate, and while that was being layed down on one track, I was using the 815 in foreground feeding another track. Because the 815 is so directional, I was able to find an angle where the mike would reject all those background sounds from the other action, and we kept the tracks isolated."

There were times on this picture, however, that Webb and McLaughlin opted for a single mike in the perspective style rather than a number of convenitonal mikes or the traditional problem solvers, the RF/lavalieres This was in spite of the difficult situations that McLaughlin often found himself in as a result. In a scene in an editor's office in the Post's news room, Jacon Robards is pacing the floor discussing a critical story with Jack Warden, Martin Balsam, Redford, and Hoffman. Again, the lighting dictated that the scene be miked from below with a 415.

"You've got all these people crossing," explains Webb, "which wouldn't have been a problem from overhead, but underneath, he had to dodge all those legs, not bump the microphone, and still get the lines."

"I literally stuck the mike between Jason's legs to get one line," adds McLaughlin, "and then pulled it out before he took another step."

Phone Hook-Ups

A less physically taxing problem on the picture was the live phone hook-ups, desired by the film's producers, from the pre-production stage.

The intent was to allow for the interaction of the actors by phone, but the off-camera voice had to be isolated on the line. Consequently, the transmitting elements of the phones on camera were removed. The actors on the set were heard by their off-camera counterparts via headsets fed from the mixing console. The off-camera actors spoke into their phone transmitters and were heard by the actors on the set via the phone receivers. The on-camera actors were miked by McLaughlin, while the off-camera actors were miked both conventionally and through the phone lines. The phone line miking was accomplished with a "Mini-Mike" attached about ½-inch from a phone receiver, which was wrapped and isolated in a case on a quiet part of the stage. This gave the editors three tracks of sound to deal with on a two person phone call: one conventionally recorded track of the on-camera actor, a similar track of the off-camera actor, and a phone line track of the latter as well. This last channel was the predominantly used audio for the off-camera players.

"What I really liked about that technique," says Webb, "is that we not only got good effect, but a better performance from the actors because they were really talking. In the scene where Redford was going from phone line to phone line with all those people on hold, it was a terrific performance, and I feel to a large part because he was getting caught up in it because he had real people on those lines. It generated an energy in the scene that I don't think would have been there had he been trying to fake it."

The Working Method

McLaughlin and Webb have worked together on 16 pictures to date and have developed a method of working that serves them both artistically and logistically.

"The element that I think is most important to our success," says McLaughlin, "is the fact that we perform two specifically different tasks to achieve the one end."

"We agree on a style up front," adds Webb, "and then go to our jobs to achieve that style. I listen primarily to the backgrounds for consistency and noise so that Chris can get in with the mike and capture the dialogue within this mood. I listen for the things that may interfere with what he's doing, things that he may not hear because he's concentrating on the specifics of the microphone and the dialogue at that moment."

"The key is to keep it consistent within the style that you've set," says McLaughlin.

"There's a direct analogy to lighting here," concludes Webb, "both in the need for consistency and in the end effect. Sound can underscore the visual elements of a film to help create the mood so that the two different methods of communication really become one in the resulting motion picture. Our job is to deliver the audio portion of that whole." □□□

Constructing Loudspeaker Systems With Predictable Preformance

Jeffrey N. White and
Raymond J. Newman

In this article we will discuss some information that should be of help in constructing vented box format low frequency sections of loudspeaker systems. The information is in a sense "shorthand" that bypasses most of the agonies and detailed considerations that go into the design of a commercially produced system in order to get directly at the realization of some practically useable low frequency reproducers.

It is important to realize that a system constructor who is working with available loudspeakers is, of necessity, dealing in a take-what-you-can-get world. He must work with loudspeakers he already has or those which can be obtained "off-the-shelf," and in this context is limited to the creation of systems which these loudspeakers will enable him to concoct. It is especially important to realize that in a broad sense a system is a careful and (hopefully) knowledgeable **coordination** of loudspeaker and enclosure that achieves certain **pre-determined** performance goals. These goals are concerned with low frequency limits, efficiency levels, system size, and the amount of acoustic output available at acceptable distortion levels. They usually require a designer to create a loudspeaker from scratch that will coordinate properly with its enclosure so as to achieve these pre-determined goals. This can be considerably more difficult and complex than working with an already given loudspeaker. We think it is important to keep this distinction in mind and appreciate that the shorthand methods being described in this article are not a complete description of the complex art of loudspeaker system design. The methods will, however, enable a constructor to achieve designs which more nearly realize the potential inherent in a given loudspeaker than would be possible through "cut and try" attempts.

The material to follow is arranged in a number of sections starting with background remarks and proceeding through design calculations and vent determination. Before proceeding into construction details of the box and grille, some important remarks are included by Raymond Newman on the often neglected subject of system output capabilities. The article concludes with a specific design example.

First Some Background

In the period of roughly 1930 to 1960, many companies and individuals were constructing systems of the sealed (acoustic suspension) type and vented (bass reflex, tuned port, phase inverter) type without much understanding of the relationship between the speaker and the cabinet it was used in. Sealed systems were fairly popular due to their simple operation and ability to reproduce low frequencies. Vented systems were more complicated and constructors were usually not able to accurately predict the performance. A typical method of constructing a vented system in the early 50s and 60s was to purchase the best woofer money could buy, with the biggest magnet and lowest free air resonance, build the biggest box the listening area would allow (anything smaller than a refrigerator was frowned upon), tune the box to the free air resonant frequency of the woofer, and enjoy perfection. Many people probably did enjoy perfection because they probably had never heard anything better. Most of those early systems were devoid of fundamental bass below 50 to 60 Hz. Many of these systems had a typical bump found in the upper bass region due to the improper coordination of loudspeaker and

The Authors

Raymond J. Newman was born in Wyandotte, Michigan, in 1938. He received a B.S.E.E. degree from the University of Michigan in 1960. From 1960 to 1962 he was employed by the Aeronutronic Division of the Ford Motor Company in Newport Beach, California, where he was responsible for the checkout and installation of instrumentation used on high-altitude research rockets and later for research and design tasks associated with the prediction and manipulation of electromagnetic fields scattered by arbitrary objects (radar cross-section analysis). From 1962 to 1967 he was employed by the Conduction Corporation, of Ann Arbor, Michigan, as a member of the Senior Analysis Staff.

Since 1967 Mr. Newman has been with Electro-Voice, of Buchanan, Michigan, serving initially in the capacity of senior engineer in charge of loudspeaker systems, and later as chief engineer of loudspeakers. He is engaged in research, design, and development of loudspeaker systems for commercial and home usage, and was an early proponent of the design concepts of Thiele and Small. Mr. Newman is a member of the AES and the IEEE. He is one of the authors of "Methods of Radar Cross-Sectional Analysis."

Jeffrey N. White was born in Jeffersonville, Indiana, in 1952. He received the B.S.E.E.T. degree from Purdue University in 1974. From 1974 to 1976 he was employed by Jeff Boat in Jeffersonville, Indiana, where he was responsible for electrical engineering of power and communications systems used on river-going towboats, barges, and most recently, the new Mississippi Queen passenger steamboat.

Since 1976, Mr. White has been with Electro-Voice, of Buchanan, Michigan, serving as a project engineer of loudspeaker systems.

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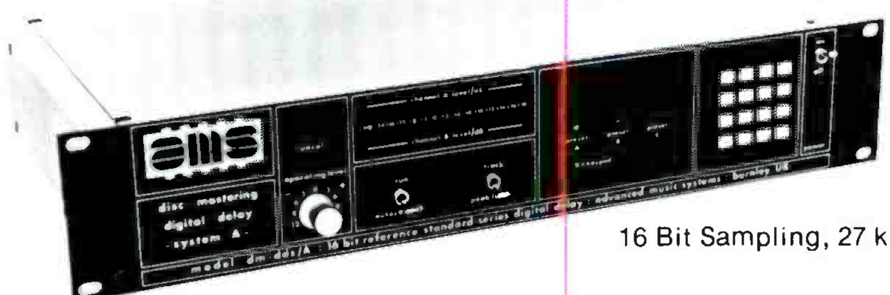

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box characteristics. Thus, the term "boom box" was given to many of the early vented systems. A few old systems did perform reasonably well — unfortunately, no one was quite sure why.

Enter The Australians

In 1961 an Australian researcher named Thiele presented the world with a mathematical analysis of low frequency speaker performance. In his article, Thiele investigates the behavior of vented box systems by analyzing their equivalent circuits as high-pass filters. Using techniques of electrical network analysis, he arrives at a number of different ways of creating vented-box speaker systems, and presents them in a table format called the "alignment table." This allows the constructor to pick the desired response before actually constructing the system and with excellent results. Not until 1971 did Thiele's work surface domestically when published in the **Journal of the Audio Engineering Society**.[1] Thiele points out that optimization is possible! There are three system factors which are interrelated for both sealed and vented systems:

(1) Low frequency limit: f_3 (where the response is 3 dB down and falling rapidly below that frequency);
(2) Efficiency (how much of the electrical input power from the amplifier is converted to acoustic output power, which is what you hear);
(3) Enclosure size (how much volume there is inside the box).

It turns out that, if any two parameters are chosen, the remaining one is determined. A constructor can juggle the three interrelationships to achieve the desired goal using Thiele's work. Thiele also demonstrated that vented boxes are not inherently high efficiency only, but can cover a wide range of efficiencies depending upon other design choices. In comparison to flat response sealed systems, vented systems offer a possibility of one of three advantages, or an appropriate mixture of all three.

(1) 1/3 octave more bass. That is like moving from 40 Hz to 32 Hz, or
(2) 4.2 dB more efficiency. That is like multiplying your amplifier power by a bit more than 2-1/2 times, or
(3) Reduce the enclosure to almost 1/3 the original size.

The work of another Australian, Dr. Richard Small, included a more detailed look into both sealed and vented systems. He discussed the process of constructing speakers to obtain parameters to fit a desired alignment. Small investigated speaker performance under large signal conditions and the effect of losses on system performance.

Alignment Details		Box Design			Aux. Ckts. Required	
No.	Type	f_3/f_s	f_s/f_b	V_{as}/V_b	Q_L	
1	QB_3	2.68	1.34	10.48	.180	no
2	QB_3	2.28	1.32	7.48	.209	no
3	QB_3	1.77	1.25	4.46	.259	no
4	QB_3	1.45	1.18	2.95	.303	no
5	B_4	1.000	1.000	1.414	.383	no
6	C_4	.867	.935	1.055	.415	no
7	C_4	.729	.879	.729	.466	no
8	C_4	.641	.847	.559	.518	no
9	C_4	.600	.838	.485	.557	no
10	B_5	1.000	1.000	1.000	.447	yes
11	C_5	.852	.934	.583	.545	yes
12	C_5	.724	.889	.273	.810	yes
13	C_5	.704	.882	.227	.924	yes
14	C_5	.685	.877	.191	1.102	yes
15	B_6	1.000	1.000	2.73	.299	yes
16	C_6	.850	.868	2.33	.317	yes
17	C_6	.698	.750	1.81	.348	yes
18	C_6	.620	.698	1.51	.371	yes
19	C_6	.554	.659	1.25	.399	yes
20	B_6	1.000	1.000	1.000	.408	yes
21	C_6	.844	.954	.722	.431	yes
22	C_6	.677	.917	.500	.461	yes
23	C_6	.592	.902	.414	.484	yes
24	C_6	.520	.890	.353	.513	yes
25	C_6	.404	.876	.276	.616	yes
26	B_6	1.000	1.000	.732	.518	yes
27	C_6	.778	.911	.110	1.503	yes
28	QB_3	.952	.980	1.89	.328	yes

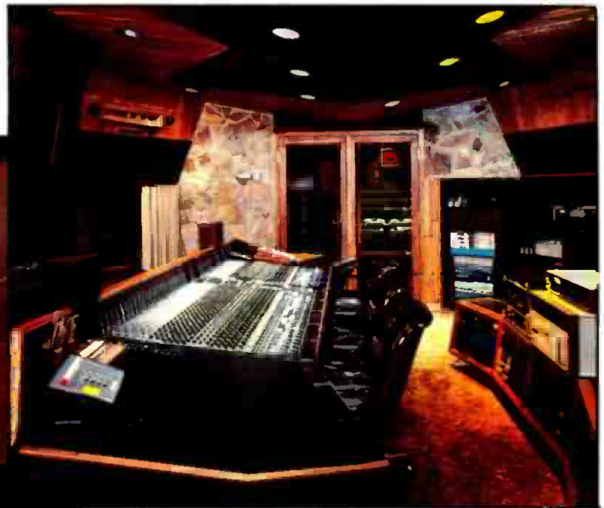
TABLE 1: Summary of Vented-Box Alignments, after Thiele.

Calculating The Design

A loudspeaker system designer armed with Thiele's equations can construct a cone loudspeaker to fit a selected response alignment from Thiele's alignment table (more on that shortly), or as is usually the case for a layman, design a system for a given loudspeaker. In this article we do not intend to describe loudspeaker design, but rather to show how to use existing speakers for system construction. The following speaker parameters must be obtained from the speaker manufacturer or by experimentally measuring them:[2]

. . . continued on page 68 —

Electric Lady Studio



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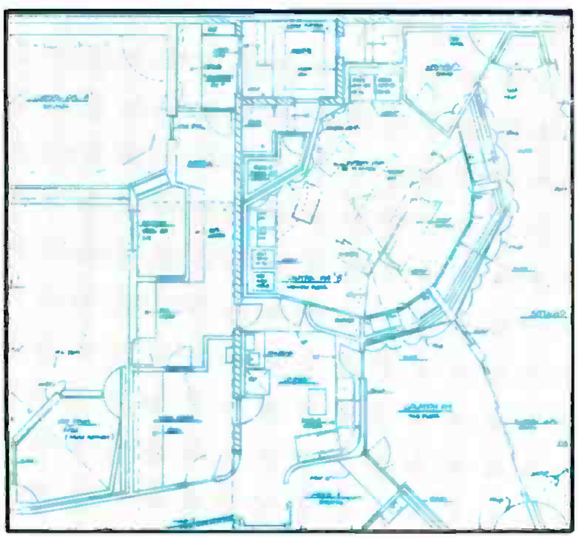

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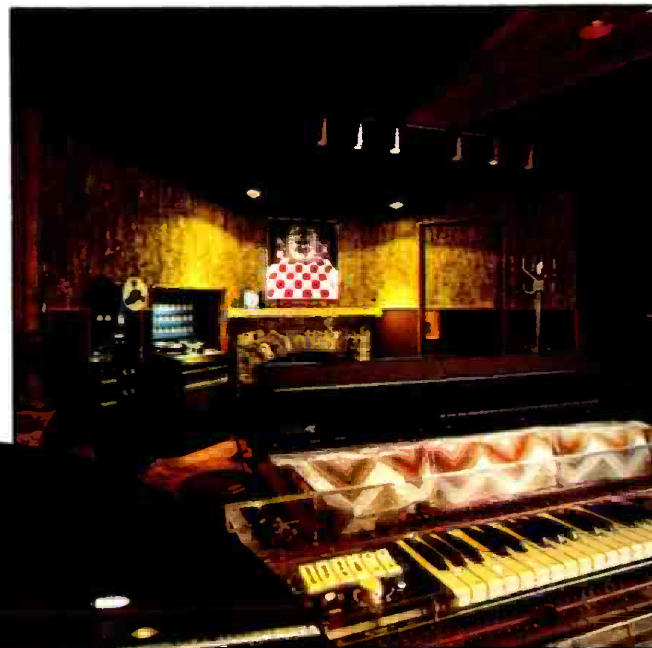

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— continued from page 64 . .

(1) Free air resonance frequency: f_s (in Hertz.)
(2) Total driver Q: Q_t (the speaker's "Q" when connected to the driving amplifier. This may be thought of as how far down the speaker's response is at its free-air resonant frequency relative to its mid-band response in a very large baffle — for instance, a Q_t = 0.5 means that the response is down 6 dB at the free air resonant frequency.
(3) The volume of air having the same acoustic compliance as the driver suspension: V_{as}.

Once these parameters are known, you will need to consult the alignment table to determine the system parameters. Table 1 is a simplified form of Thiele's table which we will use to determine system design. Probably the first question is what alignment do we use? A little explanation will be of help here. The various types of filters are shown under type as B, C, QB, etc. Each of these is an abbreviation for the type of filter represented — for example, QB_3 is a quasi-Butterworth third order filter; B is a Butterworth filter and C is a Chebyshev filter. In a nutshell, Butterworth filters are characterized by flat frequency response and the Chebyshev filters are characterized by small ripples in the response. The names are actually the names of the mathematicians who first developed the equations for the particular response shape. The subscripts can be taken to describe the rate of roll-off below the low-frequency limit. For example, the complete description of "B_4" is "fourth-order Butterworth response" and the roll-off is (6 x 4) = 24 dB per octave. A B_6 has a (6 x 6) = 36 dB per octave roll-off. Figure 1 shows how some of these curves look. The following symbol explanations will aid in understanding the table more clearly:

f_b = the frequency of box tuning (a function of box volume, vent area, and length, and not affected by the speaker itself).
V_b = the internal volume of the box.

The question is which alignment to use. The table can be split into two categories:

(1) Alignments 1 - 9; no auxiliary electrical filter or equalizer required,
(2) Alignments 10 - 28; auxiliary electrical filter or equalizer required.

The first nine alignments can be primarily thought of as a given system in which the compliance of the loudspeaker is varied. This is not exactly true from a technical standpoint, but it is a practical way to view them. Alignment 5, the fourth-order Butterworth can be viewed as the representative central alignment of the first nine. The other 19 alignments have auxiliary filters, or equalizers, which combine loudspeaker and box performance to give the specific Butterworth or Chebyshev response. Some equalizers boost response as the low-frequency limit is approached and others cut response. Most of the equalizers, below the low-frequency limit, cut input to the speaker system resulting in the high-order responses like B_6. This cut below the low-frequency limit has the useful side effect of keeping very low-frequency signals such as turntable rumble or record-surface irregularities out of the woofer. Useless, distortion producing, large but inaudible woofer motions are therefore eliminated. Alignment number 15, the sixth-order Butterworth, is probably the most representative of the latter group.

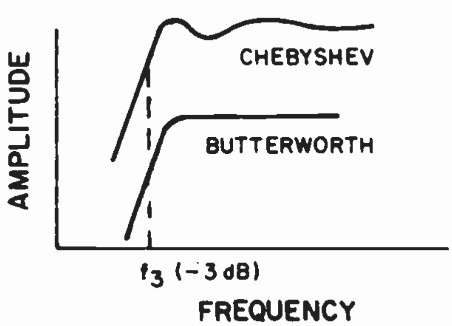


Figure 1: Nature of Butterworth and Chebyshev Filter Responses.

To calculate the design for alignment number 5, the fourth-order Butterworth, we will first need the three loudspeakers' parameters mentioned.

Thiele's alignment table tells us that a loudspeaker with a Q_t of .383 will be necessary. To find the box volume (V_b) required, the following formula will be used:

$$V_b = V_{as} \text{ divided by } 1.414$$

It should be noted that Thiele deals with lossless systems, so a factor of approximately 1.3 is included in the calculation to make up for losses. Therefore, the approximate box volume for the constructed enclosure should be the following formula:

$$V_b \text{ (constructed)} \cong (V_b \text{ lossless}) (1.3)$$

(Note: The actual box constructed should allow approximately 10% extra volume for speaker displacement and internal bracing.)

For the fourth-order Butterworth response, the box resonance frequency, or f_b, equals the f_3, or the 3 dB down, frequency of the system. The f_3, from the alignment table, happens to be f_s, or free air resonant frequency of the loudspeaker which was chosen. Therefore, the box tuning frequency and its 3 dB down point are the same as the free air resonance frequency of the speaker. This may sound like the old method of tuning the box to the free air resonance of the speaker, but realize that it is very important to know the two other parameters V_{as} and Q_t, which are interrelated in the whole system. If either of these two parameters are different than those designated by Thiele's alignment table, the response will not be flat as predicted. So by now, if you had a speaker with a Q_t of .383, you could have calculated the box's desired volume and predicted the low frequency cutoff of the system. As is more likely the case, speakers that you may possses do not have a Q_t of .383 and you are wondering if you will have to use a different alignment. Authors such as Small[3] and Keele[4] have shown that loudspeaker response is less sensitive to shifts in suspension compliance than to almost any other variable. This phenomenom can be put to use in altering other parameters of given loudspeakers to desired values. If this is done to extremes, appreciable response errors can be generated.[5] Shifting the suspension compliance also shifts the f_s, V_{as}, and Q_t of a given loudspeaker. What this amounts to is that we can mathematically scale the speaker parameters to the desired value and check a compliance error table to see if this is tolerable. If it is, we can proceed with the design. This allows the constructor to nearly always use any loudspeaker in something similar to a fourth-order Butterworth alignment and determine the performance that the optimum box would produce. In our experimental section we will go through a speaker scaling example so you can see how this works.

The Ballpark Rule

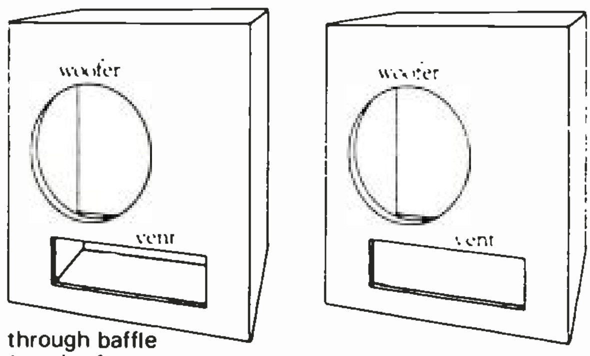
Once you have calculated the design, how much error can be tolerated? Good question! This may be important if you already have a box or space a box would fit into. Variations of box volume and box tuning frequency will cause the response curve to suffer. If the box is smaller than optimum, or the box tuning frequency is lower than optimum, the response will roll off at a higher frequency and produce an f_3 higher than the one predicted by an optimum design. By the same token, higher tunings and larger boxes will create humps in the response. Deviations of ±20% for V_b and ±10% for f_b are fairly tolerable. Later on we will show a formula that will allow you to calculate the frequency response of your own particular design. These variations let you know when you are in the ballpark for the design.

— continued overleaf . . .

Vent Design

Looking at Figure 2, the vent can be either a hole through a panel of the enclosure or a hole with some length of tube added to it. The area of the vent will be S_v or the vent cross-sectional area and the length of the vent will be L_v. Since the air just outside each end of the vent moves, in effect adding some length to the actual vent, this must be taken into account when calculating the vent length and will be designated at L_{ve} or vent effective length. In general, as the cross-sectional area is increased the box tuning (f_b) will go up if all other dimensions are held constant. As vent length is increased f_b will go down. From these two statements you can see that many combinations of vent area and length will yield the same f_b. How to pick a size? The only criterion is the minimum size that will not produce turbulent airflow noises (air rushing through a small hole so fast that it makes a disturbing sound). In many cases something on the order of three to four inches in diameter will be a good starting point.

Figure 2: Vented Boxes Showing Vent Construction Methods.



Hole through baffle with length of duct behind.

Hole through baffle.

The resonance frequency of a Helmholtz resonator (vented box)[6] is given by the following formula:

$$f_b = [C/2\pi]\,[S_v/(L_{ve} \cdot V_b)]^{1/2}$$

Where C = velocity of sound in air (343 meters/sec) or (13504 inches/sec), Keele[7] provides a neat little formula for finding the vent length.

Compute α Vent Area/Vent Effective Length

$$\alpha = V_b\,(2\pi f_b/C)^2 = 3.7 \times 10^{-4}\,V_b(\mathrm{ft}^3) \cdot f_b^2\ (\mathrm{Hertz})$$

$$L_v\ (\mathrm{Inches}) = [S_v\ (\mathrm{in})/\alpha] - .83\sqrt{S_v}$$

(.83 for tube vents, use .958 for hole through baffle)

Since the calculation may not be exact in a real world application, an extra 10 to 20% should be added on to the length so the vent can be shortened to the appropriate length to produce the tuning required. The following formula will be of help in determining the amount of vent shortening required:

$$\Delta L_v = -\Delta f_b\ 2L_v/f_b$$

where

ΔL_v = required change in vent length in inches (negative value means reduce length, which will raise the f_b)

Δf_b = f required − f actual

L_v, f_b = the initial vent length and initial box tuning frequency

The vent can be made by either constructing a box type vent from wood, or use of material such as PVC household water drain pipe. If desired, two pieces of pipe could be used to create the proper cross-sectional area while the length is maintained from calculations. For instance, a vent calculated for S_v = 25.1 square inch, and L_v = 3 inches long can be implemented with two 4 inch diameter tubes 3 inches long. Each 4 inch tube has a S_v = 12.5 square inches.

Before proceeding on with the nitty-gritty of actual construction details, it is worthwhile to stop and consider the subject of what kind of low frequency output potential might be expected from the system being labored over.

The Matter Of System Output Capabilities

A two per cent efficient system capable of withstanding an electrical input of 100 watts will produce an acoustic output of two watts — right? The answer is — not necessarily at all, unless the system is specifically designed to do this.

This statement is intended to point out one of the least understood and appreciated matters in system design — a matter which should become of increasing importance with increased demands for more extended and higher output low frequency reproduction. The increased use of digital recording techniques will almost surely force a greater understanding of the implications of system output on diaphragm or cone movement on system constructors and designers.

Although many of the popularized discussions of the Thiele and Small criteria for system design deal extensively with obtaining a desired system response characteristic (referred to in the jargon of the trade as a "small signal condition"), few of them delve extensively into the matter of excursion as a function of desired maximum output (a "large signal condition"). This is not because Thiele and Small ignored this subject. It may, perhaps, be due to not noticing the result of running out of excursion (because systems are not being pressed to their limits by available program material) or to blaming the problem on the amplifier. It may also be due to simply not getting around to thinking in detail about this fundamental concept. Let us try and advance to the heart of the matter with a few statements and an example.

Consider these statements.

(1) Given the knowledge available today (and a sensitive and savvy designer), it is possible to extend bass response with relative ease as long as high output requirements are not taxing (i.e., 20 Hz from an eight-inch loudspeaker is not difficult as long as the output level is fairly modest).

(2) High system efficiency and/or high input power capacity do not guarantee high output at low frequencies.

(3) Diaphragm movement to achieve a given level of acoustic output is a function of the frequencies being reproduced, the size of the diaphragm and the type of system being employed. It is not related to system efficiency unless the loudspeaker is burned out electrically in trying to achieve the desired diaphragm movement.

Here is an example that deals directly with the problem. The example assumes the need for an **acoustic** output of one watt from a system in a design phase. In a room of 3,000 to 4000 cubic feet of volume and average absorption, this would generate a sound pressure level approaching 115 dB in the reverberant field of the room (typically 10 to 15 feet back from the loudspeakers). Larger rooms of similar absorptive properties would reduce this level in appoximate proportion to their volume. This example (as illustrated in Table 2) assumes that loudspeakers of various sizes are placed in sealed or vented box type systems radiating into an acoustic "half space." (That is, they are acoustically close to a large wall or on a large floor.) In examining the information contained in the table, keep in mind that it is very difficult to design a loudspeaker with a **linear** total excursion much over one-half of an inch and that many units (especially inexpensive ones) may be considerably under this value. In any case, the manufacturer is the best source for obtaining information about **linear** excursion capabilities of a given loudspeaker.

These numbers should be pretty sobering — especially so if more than one watt is needed (say, a large room) and if reproduction to fairly low frequencies is required (30 Hz or below). Obviously it does little good if, say, an eight inch unit has an efficiency of 3% and the ability to handle 35 watts input at mid-band frequencies — at 20 Hz it can only give up trying to produce one watt output. It should be emphasized once more

— *continued overleaf . . .*

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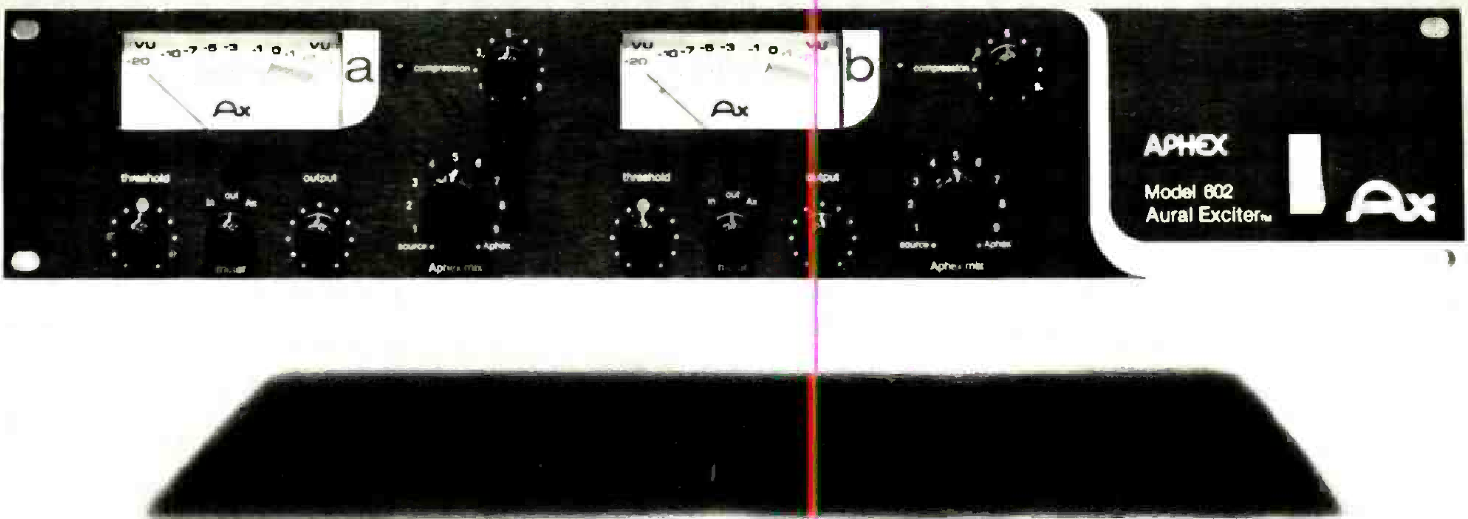

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that the excursions under consideration are those that can be reached with reasonably low distortion levels and are not those sometimes quoted as mechanical limits. (The two can differ by factors of two to four or more and, after all, what good is a fictitious output of one watt when 50% or more of output is distortion due to exceeding linear excursion limits.) What can be done about this situation?

Normal Loudspeaker Size in Inches	Required Total Passband Excusion (Inches) Needed to Reproduce Frequencies Down to Those Indicated — One Watt Output			
	50 Hz	40 Hz	30 Hz	20 Hz
8	77 (2 3)	1.2 (3.6)	2.1 (6.3)	4.8 (14.4)
10	.5 (1 5)	.78 (2.3)	1.4 (4.2)	3.1 (9.3)
12	.31 (.9)	.48 (1.4)	.86 (2.6)	1.9 (5.7)
15	.19 (.6)	.30 (.9)	.53 (1.6)	1.2 (3.6)
18	.13 (.4)	.20 (.6)	.36 (1.1)	.81 (2.4)

Table 2. Required total (peak to peak) excursion for one watt output in a passband beginning at the indicated frequency for vented and (sealed) Butterworth type response systems radiating into a half-space (For 1/4 watt output divide excursions by 2, for 4 watt output multiply by 2.)

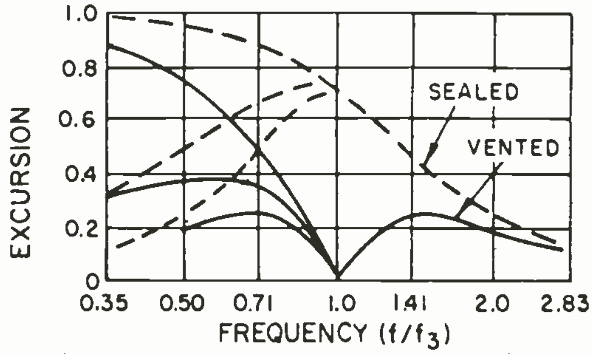

Figure 3: Relative cone excursion as a function of frequency for Butterworth-aligned vented systems and for sealed systems, with and without filters (after Thiele). Curves with reduced excursion below f/f_3 use 1st and 2nd order electrical filters.

What To Do About Inadequate Excursion

Assuming all avenues were to be open, basically five things can be done.

(1) Keep peak acoustic levels considerably below one acoustic watt (but see the digression in the sidebar).
(2) Design around a larger diameter loudspeaker (or multiple loudspeakers).
(3) Don't design around as low a bass limit as you may have originally wanted.
(4) Design the system around an environment which is less than a half-space. (A quarter-space or wall/floor juncture would reduce required excursions by 0.7, for instance.)
(5) Be sure to use a system type which minimizes excursion requirements for a given output.

The first four conditions are at least reasonably obvious once the implications of high output on excursion are realized — the last is not quite so obvious, although the information in Table 2 gives a hint as to the influence of system type. Let us discuss the excursion of certain forms of vented boxes in a bit more detail.

These "certain forms" chiefly consist of the "Butterworth" type alignments which are best characterized in this discussion as systems with reasonably flat response whose f_3's (or 3 dB down frequencies) are at or near the f_b or tuning frequency of the box they are in. These are the types that have been discussed previously. Figure 3 from Thiele's original work [1, page 473] indicates the situation. Several points of interest may be gotten from an examination of this figure.

(1) Unlike the sealed box, the frequency of peak excursion occurs approximately one-half octave above the 3 dB down point of the system.
(2) The maximum value of the excursion for a given output is approximately 1/3 that of the sealed system — even for Butterworth systems that require auxiliary circuits or equalizers as a part of their makeup.
(3) Electrical cut-off of output in the systems sub-passband (below its f_3) can greatly aid in limiting sub-passband excursions.

Assuming that it is evident that system type can have a substantial influence on the excursion-output situation, what can traditionally be done about it once a system design is arrived at based on the earlier information in this article? (Recall, incidentally, that Table 2 is for a single system. If it is presumed that a stereo pair exists and that bass output is additive, the excursions for a single system in the table can be reduced by .7.)

One situation is just to appreciate where you are and not expect the system that has been designed from available components to do much more. (Some degree of system caused clipping, perhaps of the order of 3 dB or so, will probably not be too noticeable under many conditions.) If, however, a choice of loudspeaker diameter or **linear** excursion limit is possible, it would be desirable to consider the information in Table 2 and choose accordingly. Keep in mind that a peak output capability of 1/4 watt per system (note the table caption) may be entirely adequate for smaller rooms and many musical styles. For larger rooms and/or high intensity musical styles, one or even more watts may be desirable.

In considering a system, the designer must be concerned not only with the nature of the system's response but also with what output is needed under the conditions of usage. Ignoring this consideration will put the user at the mercy of chance, perhaps resulting in either a system which is inadequate for its purpose or one which is overdesigned at his expense.

Box Design

The enclosure builder should prepar outer dimensional drawings of the cabinet to his selected proportions. Include all styling details, such as flush or recessed grille. Typical examples are shown in Figures 4 and 5. It is suggested that you also prepare detail drawings from which sawing, drilling, and part placement can be accomplished more easily and accurately. Don't forget to allow volume for speaker volume, vent volume, and internal cabinet bracing. The mitered-corner construction shown in Figure 4 is recommended; however, the simple butt-joint shown in Figure 5 can be used. In any case, one inch square glue blocks should be a part of the cabinet design to assure rigidity and strength. The glue blocks should extend nearly the entire length of the inside corners

A COMMENT ON ACOUSTIC OUTPUT REQUIREMENTS

The information in Table 2 presumes the need for one acoustic watt of output. Since excursion varies as the square root of output ($\sqrt{W_a}$) with all other matters held constant, the need for less output eases matters. For example, 1/4 watt would reduce the table entries to 1/2 of the values listed. For further understanding we need to consider the **peak** outputs inherent in musical material. Peak is emphasized because if program material is carefully examined, it will be found that present in longer duration passages (of the order of a second or a few seconds) there will be peaks present that exceed the longer average levels by 10 to 15 dB. These peaks are caused by the attack transients of instruments such as the initial sounds from brass instruments and the first contact with a drum head. Amplifier manufacturers have recognized this condition and have, as a result, engaged in the manufacture of higher power devices to encompass these peak amplitudes. The impact of this situation on loudspeakers has, in most cases, yet to be fully realized.

Although a several second output from a symphony orchestra of 100 dB would in most cases be judged as very loud, this output would contain peak levels of as much as 110 or 115 dB. 115 dB would be associated with one acoustic watt in a 3,000 or 4,000 cubic foot room of average absorption. Larger rooms would require proportionally higher output wattages. Since some musical styles can achieve average levels exceeding 100 dB, one acoustic watt of output would seem to be an entirely reasonable **peak** level to center a discussion about. □□□

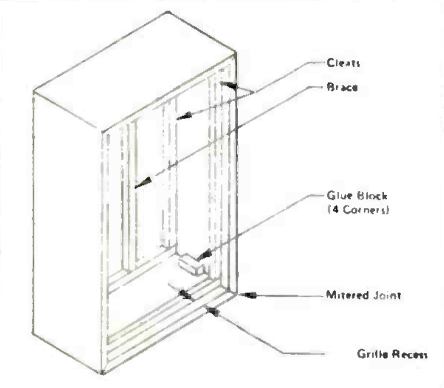

Figure 4: Recommended Mitered-Joint Box Construction.

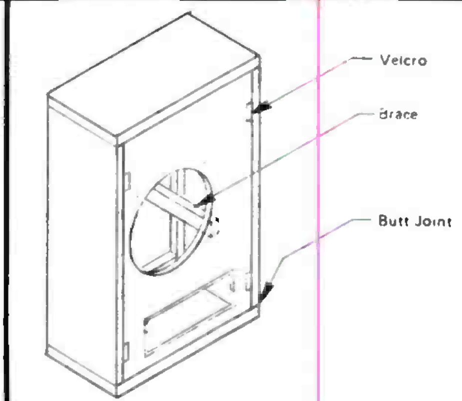

Figure 5: Alternate Butt-Joint Box Construction.

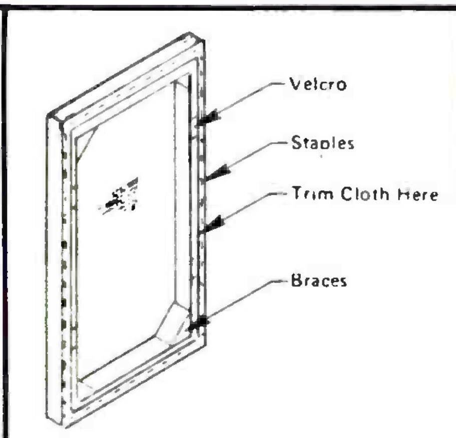

Figure 6: Back View of Grille Assembly.

that are being reinforced. These same blocks may be used for the cleats that will support both front and rear panels. If the speaker components selected are to be mounted on the back side of the front panel (from inside the cabinet), the front or rear panel must be removable in your design. Removable panels should be secured with wood screws and weather stripping tape. Notice that Figure 4 provides for a recessed grille that can be flush with the front of the cabinet, while Figure 5 provides for a grille that will project from the front of the cabinet.

The grille framework (see Figure 6) may also be constructed of one inch square wood; however, this framework will require braces if it becomes very large. You can use any height, width, or depth, provided that cabinet volume is not changed. It is important, however, that the proportions should not be extreme. Long, narrow cabinets can cause particular problems. We recommend that the width be approximately two-thirds the height and the depth be approximately one-third the height. (The vent opening inside the box should not be too near the rear to avoid constricting airflow. A clearance equal to the length of the shortest side of the vent will usually do.) Although considerable variation can be tolerated, the shortest dimension of the cabinet is best kept less than one-third that of the longest dimension.

Where you put the speakers on the front of the cabinet does not affect deep bass response, but close-to-ear-level mounting will provide the best mid- and high-frequency performance. Keep the speakers and the vent at least two inches away from the inside corners and edges of the enclosure. The vent is usually placed on the front of the cabinet, but it can be on any other side as long as it will not be blocked off in use. Figure 5 is a typical example of where speakers and vents may be placed.

CAUTION should be exercised when locating the speakers on the front panel to avoid interference with the front panel mounting cleats (see Figure 4), or interference with the grille

— continued overleaf . . .

frame that will be placed over the front panel. Also, any braces on the grille frame should not interfere with the speakers.

The width and height dimensional proportions of the vent are essentially non-critical (they can even be circular), as long as the vent cross-sectional area (S_v) is maintained. However, it is suggested that one dimension be no more than five times the other, as a narrow vent may cause excessive acoustic losses.

Material Selection

It is recommended that 3/4" thick plywood or particle board, veneered if desired, be used for all six cabinet sides, although smaller enclosures (approximately two cubic feet and under) may be successfully constructed of 5/8" material. A one inch square pine or birch strip is recommended for corner reinforcing strips (glue blocks) and the strips that form a support to attach front and rear panels (cleats).

It is recommended that the grille cloth material be any fabric or material that is specially designed for speaker grille use. Fuzzy materials should be avoided.

Fabrication And Cabinet Assembly

Lay out all cabinet sides on the sheets of plywood or particle board that will be used, per your dimensional drawings, to minimize material waste. Speaker components can be used as a template to locate their mounting holes in the front panel.

All cabinet joints must be securely glued together to assure a strong construction. "Elmer's" type (polyvinyl) wood glues are satisfactory for indoor use. All wood cleats must be securely glued and nailed or screwed down into position. The nails or screws through the cleats will retain all wood parts until the glue sets up. The non-removable front and rear panels must also be securely glued and nailed into position. The front and rear panels may be used to square up your cabinet during assembly.

In the largest boxes — greater than about six cubic feet — bracing is usually required for the largest expanses of wood to prevent sympathetic vibrations from affecting overall system performance. Proper bracing technique splits a rectangular panel into two equal rectangles with the brace placed along the panel's longest dimension. Good bracing materials are 2" x 2" dimensions lumber or four inch width of 3/4-inch plywood, placed on edge, 3/4" x 3/4", or 1" x 1" material may also be used (see Figure 4). Also, on larger cabinets it is recommended that a brace be glued between the front and rear panels, as near to the center of the panels as possible without interferring with the speakers (see Figure 5).

After all glue has dried, apply a bead of sealer (such as G.E. or Dow Corning adhesive/sealant) to all internal joints to assure an airtight cabinet.

Grille Assembly

Securely glue and nail the wood grille frame together. If additional bracing is added, be certain that it does not interfere with speakers. Paint the wood frame with a color that will not show through the grille cloth.

The grille cloth should be stretched around the frame and stapled to the rear side (see Figure 6). Velcro fasteners are handy for retaining your grille. Attach them to the back of the grille frame and to the corresponding location on the front panel of the cabinet. If so desired, the grille assembly can be fastened to the cabinet with screws.

Speaker Installation

Install crossover (if one is to be used) and terminals. Any resultant holes through cabinet walls should be sealed.

Three mutually adjacent inside surfaces of the enclosure (top, one side, and rear) should be lined with a one- or two-inch thickness of glass wool or similar acoustic absorptive material to prevent internal reflections from affecting mid-frequency performance. No absorptive material should be placed over or within the port.

Polyurethane foam tape should be used between the back panel and mounting cleats if this panel is to be removable, or between the speakers and baffle if the loudspeakers are front mounted to provide an air seal.

Sample Design

A 15" loudspeaker was determined to possess the following Thiele parameters:

f_s = 32 Hertz
Q_t = .306
V_{as} = 10.86 Cubic Feet

We would like to use this speaker for a fourth-order alignment which requires a Q_t of .383. Q_t, f_s, and V_{as} may be shifted in a systematic way if the compliance of the loudspeaker were to be hypothetically altered. If the shift is done to obtain a desired Q_t, it goes as follows:

For Q_t = .383

$$f_s' = [Q_t\ (\text{desired})/Q_t\ (\text{actual})]\ f_s\ (\text{actual}) = [.383/.306] \cdot (32) = 40 \text{ Hertz}$$

$$V_{as}' = [Q_t\ (\text{actual})/Q_t\ (\text{desired})]^2\ V_{as}\ (\text{actual}) = [.306/.383]^2 \cdot (10.86) = 6.93 \text{ Cu. Ft.}$$

What has been done is to hypothetically shift compliance until the loudspeaker is a perfect candidate for the fourth-order Butterworth. When this is done the effect of the error accummulated by not **actually** shifting the compliance can be assessed in the following fashion:

$$C_{as}\ (\text{error}) = V_{as}\ (\text{error}) = V_{as}\ (\text{actual})/V_{as}\ (\text{desired}) = 10.86/6.93 = 1.57$$

Errors of .25 to 4 are fairly tolerable in this type of manipultation as they result in response changes that amount to usually less than 1 dB.

We can now proceed with the box design:

$$V_b = [(V_{as}'/1.414) \cdot (1.3)] + 10\% = [(6.93/1.414)(1.3)] + 10\% = 7.0 \text{ Cu. Ft.}$$

$$f_b = f_s' = f_3 = 40 \text{ Hertz}$$

Vent Design

S_v = 70 Square Inches

$$\alpha = 3.7 \times 10^{-4} \cdot (7)(40)^2 = 4.144$$

$$L_v = [S_v/\alpha] - .83\sqrt{S_v} = [70/4.144] - (.83)\sqrt{70} = 16.89 - 6.94 = 9.95 \text{ Inches}$$

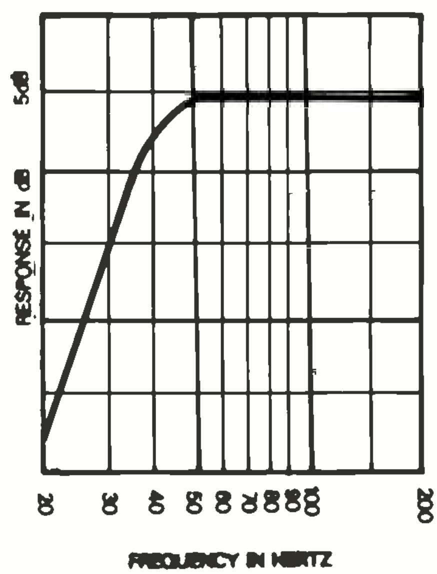

Figure 7: Low-Frequency Response of Example System f_b = 40 Hertz, f_3 = 40 Hertz, V_b = 7.0 Cu. Ft.

Adding an extra 10% makes the vent 10.95 or just 11.0 inches.

When the enclosure was constructed the F_h was 35.5 Hertz with the 11 inch long vent. Applying the following formula gives the amount to be cut off to raise the tuning to the desired 40 Hertz.

$$\Delta L_v = -\Delta f_h\ 2\ L_v/f_h = -(40\text{-}35.5)\ (2)(11)/35.5 = -2.78 \text{ Inches}$$

The vent was shortened to 8-1/4 inches to obtain a 40 Hertz tuning. Figure 7 shows the response of the 40 Hertz low frequency system.

For those people who have access to a programmable calculator or computer, the following formula can be used to calculate the response of the system design:

Constants: $A = f_h^2/f_s^2$

$$B = (A/Q_t) + [f_h/(7\ f_s)]$$

$$C = 1 + A + [f_h/(7\ f_s Q_t)] + (V_{as}/V_b)$$

$$D = (1/Q_t) + [f_h/(7\ f_s)]$$

For each frequency (f) of interest find $f_n = f/f_s$

Substitute each f_n into the following formula:

$$\text{Response (dB)} = 20 \log\ f_n^4/\sqrt{[(f_n^4 - Cf_n^2 + A)^2 + (Bf_n - Df_n^3)^2]}$$

An example is shown in Figure 8. □□□

FREQ (Hz)	RELATIVE RESPONSE (dB)	FREQ (Hz)	RELATIVE RESPONSE (dB)	FREQ (Hz)	RELATIVE RESPONSE (dB)
30.0	-10.6	54.0	-0.6	78.0	-0.5
32.0	-8.5	56.0	-0.6	80.0	-0.5
34.0	-6.6	58.0	-0.5	82.0	-0.5
36.0	-5.0	60.0	-0.5	84.0	-0.5
38.0	-3.7	62.0	-0.5	86.0	-0.5
40.0	-2.7	64.0	-0.5	88.0	-0.4
42.0	-1.9	66.0	-0.5	90.0	-0.4
44.0	-1.4	68.0	-0.5	92.0	-0.4
46.0	-1.1	70.0	-0.5	94.0	-0.4
48.0	-0.9	72.0	-0.5	96.0	-0.4
50.0	-0.7	74.0	-0.5	98.0	-0.4
52.0	-0.6	76.0	-0.5	100	-0.4

FIGURE 8: FORMULA-CALCULATED LOW FREQUENCY RESPONSE OF EXAMPLE SYSTEM

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6 - L. E. Kinsler and A. R. Frey, "Fundamentals of Acoustics," (Wiley, New York, 1962).

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SOUND MAN'S GUIDE TO VENUES

— number 11 in the series —

CONCORD PAVILION
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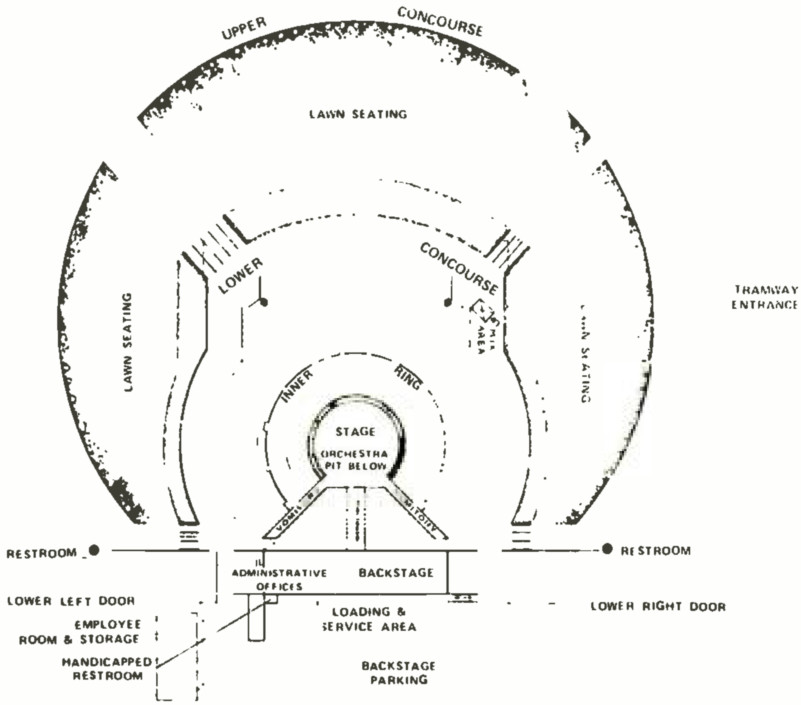
Concord, CA 94521
(415) 798-3311 - Administration
(415) 798-3319 - Production



Facility

Outdoor pavilion-type theatre with 3,500 seats under roof; 4,500 lawn seats. Open 7:30 a.m. to 4 p.m., Monday through Friday. No enforced closing time for concerts. Round stage has seating on all sides, but is generally played 3/4 in round with backdrop. Level stage is 55' in diameter. Front section of stage (a 38' x 12' crescent-shaped piece) doubles as an orchestra pit and can be lowered. First row of seating is one foot from stage apron. Grid is 22' above stage. Six pipes are available to hang monitors; will hold 500 lbs. each.

Acoustics

Good with full house; boomy if under 3/4 full but workable with use of EQ. Additional absorption planned for 1981 season. Stage acoustics a little bass heavy, partially due to backwash from overhead PA encircling stage. Very workable with good monitors — a carpet helps as well. Seats are hardback; no actual seats on lawn. Acoustics are variable through use of "Assisted Resonance System" designed by AIRO. This electronic system increases low end reverb time for symphonic applications. Stage is totally surrounded by an "acoustical moat" which can be filled in as necessary with 3' x 4' sections of staging.

Loading

14'H x 12'W loading door at rear of pavilion lowers to street level. Long aluminum ramp available to assist unloading. 50' straight hallway leads to stage. Parking for up to 10 semi's available behind theatre.

Setup

Platform (12' x 6' x 1') at console area is 90' from stage, off stage right. Cable length of 175' needed to reach center stage via approved routing. Stage level speaker stacks are unacceptable due to poor coverage and large number of blocked sightlines. Theatre has facilities and personnel to hang speakers for full 360 degree coverage if necessary. Total grid weight capacity is 8 tons. Two 40 input mike splitter boxes on stage; one 40 input box under stage. Splitter utilizes Deane Jensen JE-MB-C transformers and is patchable for direct or isolated outputs to three locations simultaneously: 1) sound booth; 2) in-house mixer; 3) stage monitor mixer. First 27 pair of pins in AMP connector mate with McCune Sound, Wally Heider, and Record Plant snakes. Ten line-feed cables run from house mix position through sound booth to stage in separate conduit. Six grid mike lines, two channel Clear-Com, telephone IC, backstage call mike, CCTV are all available at house mix position. Two 20 amp AC circuits on isolation transformer located at mix area.

Sound System

- house speakers -

Under-roof system consists of 15 Spectra Sonics Model 3000 tri-amped enclosures distributed in a ring above circumference of stage. Each cabinet is powered by a total of 4 model 700 amps driving one each JBL 2220, JBL 2420, and E-V T-350 drivers. Five additional enclosures are located 20' out from this inner ring and are on an Industrial Research time delay. This system is designed to produce 110 dB throughout inner seating area. In addition, high-power speakers are hung each season to reproduce pop and rock music. In 1980, 10 enclosures from Sound-On-Stage were used consisting of 2 - 15" Gauss woofers, 1 - JBL 2220, 1 - JBL 2440 on a Harbinger horn and either 1 or 2 JBL slot radiators.

The lawn system consists of 10 quad-amped Spectra Sonics Model 3085 cabinets and 2 Model 3000 tri-amped speakers. This system is also designed to produce 110 dB in the lawn area and is also delayed. For higher levels, 8 additional 4-way cabinets built by FM Productions are employed, each containing 6 - E-V 15LM woofers, 4 - JBL K120s, 1 - JBL 2482, and 4 - JBL 2420 drivers. This system is powered by 10 Crown DC300As and 4 Phase Linear 700s.

- house console -

Located in overhead booth and rarely used except for symphony and light jazz is a Spectra Sonics console: 20 in, 4 subs, 3 outs, 3 cue sends, 2 band EQ. Will accept balanced line level signal on male XLR connector. Yamaha PM 1000-24 or PM 2000-32 consoles are available at extra cost for mixing in the house.

- monitor system -

Monitor mixer integrated into house console. Four Altec 1221 speakers and three Spectra Sonics 3085 tri-amped speakers available. Portable monitor mix boards with up to eight discrete outputs, as well as high-power monitor speakers, are available at extra cost.

- microphones, stands, etc. -

(20) AKG C-451; (3) AKG CK-5 capsules; (3) AKG CK-9 capsules; (4) AKG D-140; (6) AKG D-190; (2) AKG 110; (2) Beyer M-500; (3) Sennheiser 421; (1) Sony ECM-50; (3) Countryman direct boxes; (1) Countryman piano pickup.

(25) Beyer mike stands; (17) Beyer boom arms; (6) goosenecks; (4) short stands.

Orban Parametric EQ; AKG BX-20 reverb; UREI feedback suppressors; one-half track and cassette machines.

There is a charge for this house sound system which varies depending on what is required for each show.

RT60 Measurements	125 Hz	250 Hz	500 Hz	1,000 Hz
Empty	2.8 sec.	2.3 sec.	2.0 sec.	1.9 sec.
3/4 Full	2.3 sec.	1.9 sec.	1.6 sec.	1.5 sec.
Empty (with ARS)	3.7 sec.	3.2 sec.	3.3 sec.	3.1 sec.
3/4 Full (with ARS)	2.8 sec.	2.5 sec.	2.3 sec.	2.1 sec.

Electrical

Two three-phase, five-wire, 400 amp services supply 2,400 amps total. Main breaker box is 60' from stage and requires lug connectors. Ten 20 amp, 118 volt circuits are available on stage. SCR lighting equipment.

Personnel

Union house, non-departmentalized. Separate crew call for truck loaders not necessary.

Building Manager - Bill Cambra, (415) 671-3287.

Telephone (415) 798-3319 for the following personnel:

Stage Manager - Doug Warrick.
Sound head - Skip Spragens.
Chief electrician - James Hussey.
Piano Tuner - James Donelson.

Traveling Soundman Reaction

"Difficult to get a good sound under the roof with loud acts due to a low mid frequency problem which is hard to deal with. They are aware of problem, however, and are planning extra sound treatment. The cavernous basement/pit area under stage may be part of the problem. Try to keep as much sound out of the ceiling as possible. Keep on-stage sound and band equipment low or you will block sightlines. Monitor mixer sets up in acoustical moat below audience sightlines. Mix position doesn't seem to be real representative of what is happening throughout the rest of the house, so move around a bit at sound check to get a better idea." ***Steve Neal**, FM Productions.*

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Bi-amping and tri-amping (electronically dividing the frequency spectrum into low-high/ low-mid-high bands and amplifying them separately) are quickly becoming the excepted methods for insuring the low levels of intermodulation and harmonic distortion so necessary for clean, accurate reproduction of today's music. Unfortunately, this involves the use of multiple power amplifiers and electronic crossover networks — usually a substantial financial burden for the average musician or sound engineer.

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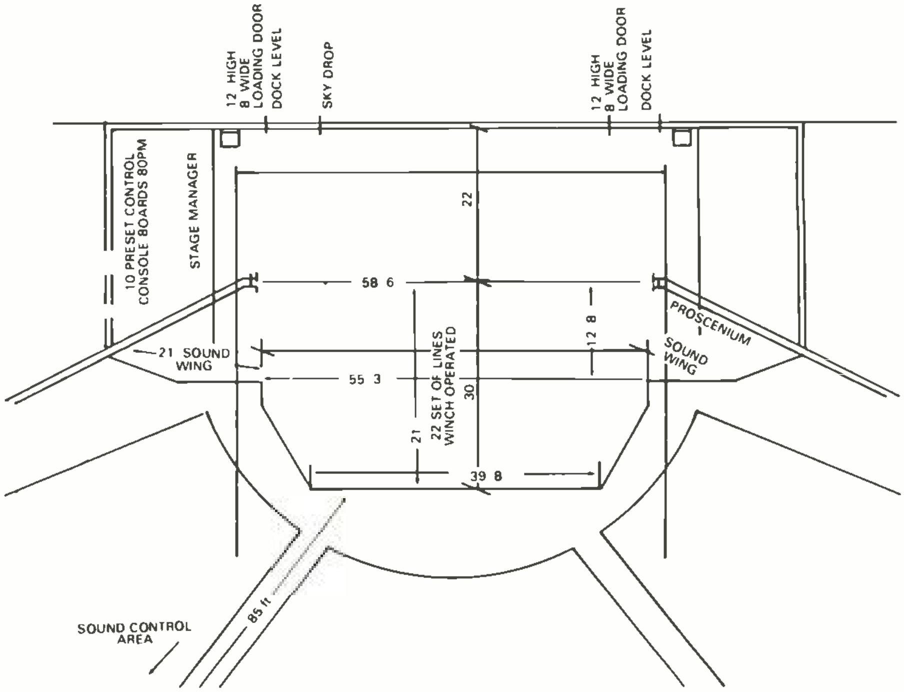
SOUND MAN'S GUIDE TO VENUES

— number 12 in the series —



PINE KNOB MUSIC THEATRE
7777 Pine Knob Road
Clarkston, Michigan 48016
(313) 625-0511
(313) 625-5250 (Backstage)

Driving Directions From Airport

From Detroit Metropolitan Airport take I-94 East to I-75 interchange. North on I-75 toward Flint, Michigan, to Sashabah Road Exit; right on Siashabah. Allow 1-¼-hour drive time from airport.

Facility

Outdoor pavilion-type theatre with 5,920 seats under roof; 4,580 lawn seats. Open normal hours; call first. Concerts must conclude no later than 11:00 p.m. No orchestra pit. Level stage thrusts into audience somewhat. Seating begins 5' from stage apron. Height from stage to grid: 45'. See diagram for stage dimensions. Some pipes available for hanging monitors.

Acoustics

Moderately boomy under roof due to corrugated metal ceiling. Sound carries to rear of covered seating area easily. Stage acoustics balanced but tend to be uncomfortably "hot" with loud rock groups. Seats all are hardback; no actual seats on lawn.

RT_{60} measurements: Low frequency = 3½ sec.; Mid frequency = 3½ sec.; High frequency = 1¼ sec.

Loading

Two loading doors (12' H x 8' W) at back of stage, loading dock level. Equipment unloads directly onto stage. Parking available for trucks at rear of theatre.

Setup

Area reserved for mixing console measures 10' x 6' and is located 85' from stage, slightly left of center. 125' cable necessary to run down aisle to center stage. Grounded AC outlet located at mixer position. Sound wings located both sides of stage in front of proscenium but behind thrust section of stage. Some sightlines blocked to rear of stage depending on size of speaker stacks. Standard orchestra risers (4' x 8') are available to build speaker platforms and are available in 8", 16", and 24" heights. Extremely difficult to fly speakers here due to location of weight bearing beams. Theatre strongly recommends against it.

Sound System

- house speakers -

Speakers are two-way, bi-amped. 24 woofers and 15 horns are divided among 3 overhead hanging clusters and produce a maximum sound level of 110 dB at the rear seats under the roof. Eight satellite clusters consisting of 2 woofers and one horn each are connected to a time delay for lawn coverage. Marantz amplifiers provide 4,200 watts for the covered seating system and 2,000 watts for the lawn speakers. All low frequency speakers are JBL 2220Bs in vented 4550 horn enclosures. All high frequency speakers are JBL 2441 drivers on either JBL 2355 or 2350 horns as required.

- house console -

Yamaha PM 1000-32. 32 mike or line inputs, 4 subs, 8 outputs, 3 band EQ with variable mid frequency. Can accept balanced line level signal on male XLR connector.

- monitor system -

On-stage portable monitor console made by FSI. 24 input, 3 discrete outs, 3 band EQ. Full monitor split box with ground lifts on 54 mike lines which terminate in individual XLR connectors. 1,200 watt monitor amp rack houses Marantz amps. Three types of monitor speakers all utilize JBL components and are all bi-amped: 4 "slopes" contain two K-120 woofers, one 2410 driver on a 2305 lens in Northwest Sound cabinets; 4 "cubes" contain one K-130 woofer, one 2470 driver on a 2345 horn; and 4 "sidefills" contain one K-140 woofer in a 4560 "Perkins" cabinet, one 2440 driver on a 2355 horn.

There is a charge of approximately $300 for the use of the monitor system and a monitor mix engineer. No charge for use of house system alone.

- microphones -

(6) Shure SM-58, (4) SM-57, (12) SM-53, (2) E-V RE-55, (6) Sony ECM-270, (4) Superscope EC-9P, (12) Superscope EC-12 tie tack mikes.

- stands and miscellaneous -

Atlas stands consisting of (12) MS-20 floor stands, (15) MS-20 with BB-1 or BB-1x booms, (6) short stands. Various goosenecks, clamps, etc. (3) SESCOM direct boxes. Overall house limiting via dbx-160. Crown EQ-2 equalizers used for house and monitor EQ.

Electrical

Three phase, 1,200 amps per leg provides 3,600 amps for sound and lights. Main breaker box is 50' from stage and requires pigtail connectors. Ten 25 amp, 110 volt circuits available on stage. SCR lighting system.

Personnel

Union house, departmentalized. Separate union crew required for truck loading.
Building Manager - Joel DeShane
Stage Manager - Jack Kovacs
Sound Chief - James Alexander
Assistant Sound - Henry Ruiz
Lighting head, Chief electrician - Pat O'Dea
Piano Tuner - Peter Berton

All available through telephone (313) 625-0511.

Traveling Soundman Reaction:

"Metal roof is not too well damped and causes a resonance between 150 - 200 Hz. Lawn system better than most. House system adequate as fill for your own PA. They hose down the place at night so get equipment up on 2 x 4's and cover anything that would be harmed by water! Mix location is slightly left of center making a stereo mix difficult. Simple load-in; very helpful crew — sound man Jim is very good. Generally enthusiastic crowds." ***Dave Morgan,*** *A-1 Audio.*

"Basic outdoor pavilion-type place. House system is OK for low level groups, but lacks power and "balls" for rock. Hard to eliminate the boominess under the roof — high end seems to drop away quickly as well. Recommend getting speakers up as high as possible and using lots of power. Load-in is great with a good efficient crew. Lawn system seemed a bit uneven in response — may be due to possible defective drivers when I was there." ***Terry Enloe,*** *McCune Sound.*

SOUNDMAN'S GUIDE to VENUES

*is a series being compiled by **R-e/p**'s sound reinforcement consulting editor, **Pat Maloney,** whose full-time profession is as an internationally recognized sound reinforcement engineer/mixer. The series is the result of a questionnaire Pat developed to be sent to performance venues in anticipation of the start of a concert tour. The information returned by the venue is considered vital to pre-planning the tour. Periodically **R-e/p** will offer an updated collection of the reports published. — ed.*

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So we moved to an unyielding slate floor. Here it took three more drops on its side from six feet, and three more on its nose from four feet to finally affect the ATM41. A truly remarkable record!

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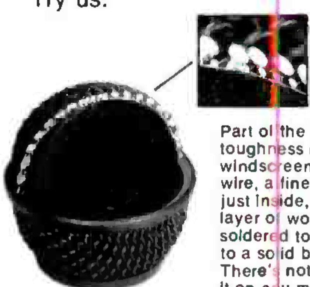
Our little test left us arm-weary but convinced that the ATM Series microphones could easily earn their "Road Tough" name in the field. That's the testing which really counts. Try us.



Part of the secret of ATM toughness is this 3-layer windscreen. An outer heavy wire, a finer wire screen just inside, and an inner layer of woven bronze. All soldered to each other and to a solid brass ring. There's nothing else like it on any microphone.

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Frank Sinatra... "Singin' in the Rain" For 140,000 at Rios' Maracana Stadium

by Patrick Maloney

When Al Siniscal, president and owner of A-1 Audio, was asked by Bob Kiernan of Sinatra Enterprises to provide the sound system for the Frank Sinatra Concert in Rio de Janeiro to be held during the summer of 1980, he readily accepted. The fact that the concert was to take place outdoors "in the round" in the 150,000 seat Maracana Soccer Stadium, made the event a challenge Al couldn't resist. After all it was only November of 1979 — which left him with at least seven months to prepare for the concert. Eminently qualified to do a show of this size, A-1 Audio is experienced in all phases of the professional audio field and enjoys a reputation for providing reliable high-quality sound systems for star performers such as The Doobie Brothers, Paul Anka, Barry Manilow, and many others. However, as Al got a little further into the planning of this show, he started encountering a few problems. Yes, the show was in the summer but summertime in Rio comes in January, not in June! The seven-month lead time shrunk to a month-and-a-half. Another problem was the fact that Brazil is situated in the tropics. It rains in the summer in the tropics — it rains every single day. In fact, the reason that the Stadium was available at all was because very few people like to play or even watch soccer in the rain. Why the promoters wanted to take the risk of doing a show with that much hazard of rain is perhaps another story. Nevertheless they did and A-1 elected to cope with it.

Coping with doing a show outdoors with an expected audience of over 140,000 people is challenging enough for any audio company. The additional fact that it was to be televised "live" throughout South America only added to the complexity. Having to do it in the round presented its own unique set of problems. But the very real possibility of doing it in the rain created an acute situation to be reckoned with!

The People

Bob Kiernan was the overall designer and production manager of the event working very closely with **Carlos Scorzelli** representing Artplan Publicidad, the Brazilian promoters. Bob, who has been with Frank Sinatra for many years, was responsible for mixing the sound as well as calling the lights, so he was in total control of the show. **Bobby Ross,** head of engineering at A-1 Audio, laid out all of the equipment and planned most of the show with the production staff that was going to Rio. The three people from A-1 who actually operated the show were Al Siniscal, Grey Ingram, and Pat Weber. **Grey Ingram** is chief engineer for the Doobie Brothers and is, according to Al, "a very knowledgeable engineer who knows electronic equipment in and out. We felt that since we were going as far away as Brazil we had to be totally self-sufficient because there were no local service stations anywhere to fix anything." Bob Kiernan specified that Al himself had to go if A-1 was going to do the show, as Al knows every resistor and wire in the system. The third person, **Pat Weber,** is presently an independent engineer who is working maintenance at Record Plant, Los Angeles, and who teaches a sound reinforcement course at the University of Sound Arts. Aside from being a qualified technical person, he also speaks Spanish and some Portuguese — the main language of Brazil. As it turned out, Pat's interpreting helped with the installation tremendously. The promoter did supply an interpreter, but since there were many tasks and activities happening concurrently there weren't enough interpreters around to get everything done in time. As a general comment about working where interpreters are necessary, they are rarely technically knowledgeable. So Pat was able to do a better job in this respect. Going to a foreign country without arranging for a full-time interpreter or bringing one yourself is a mistake you'll only make once!

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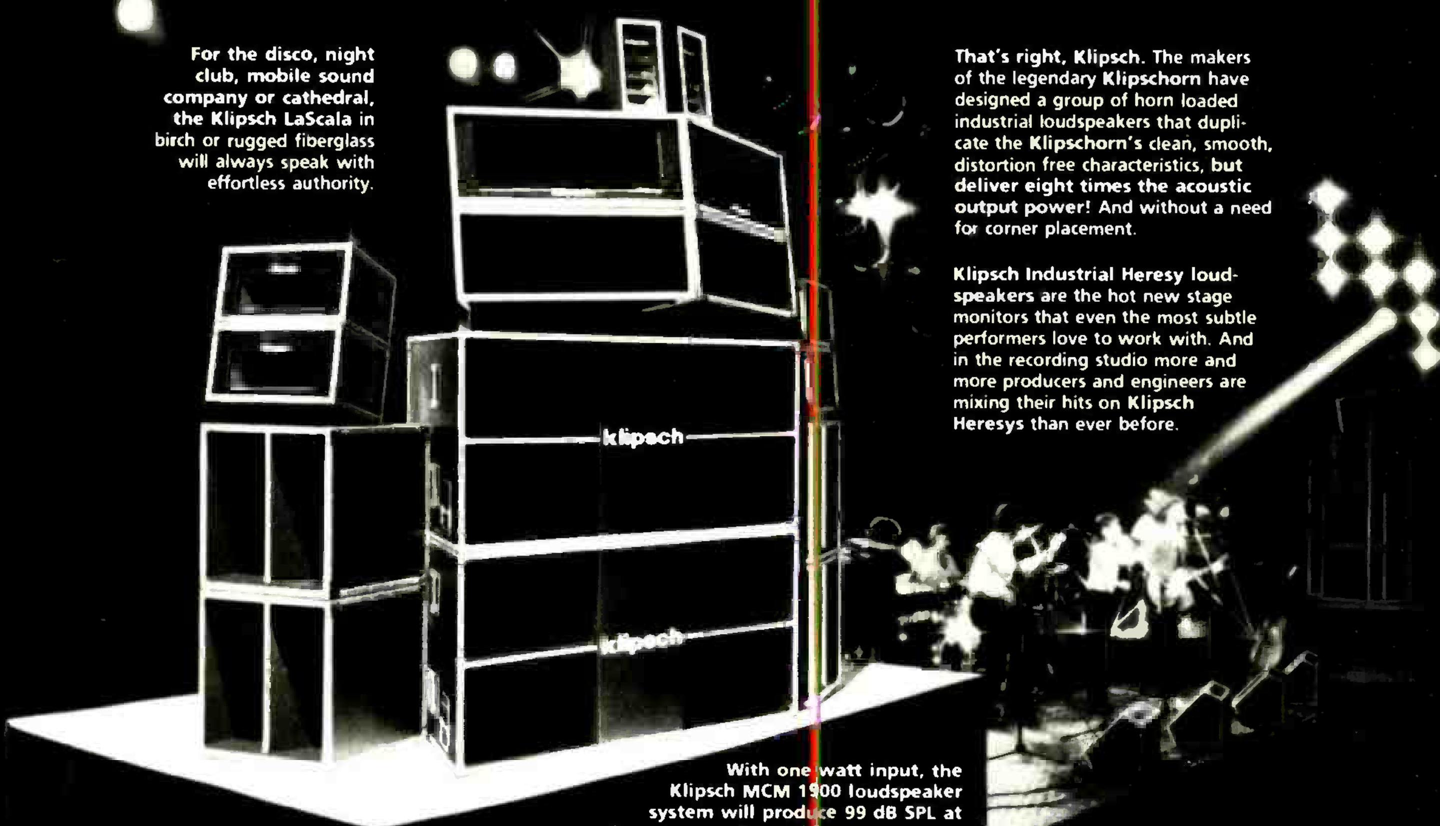

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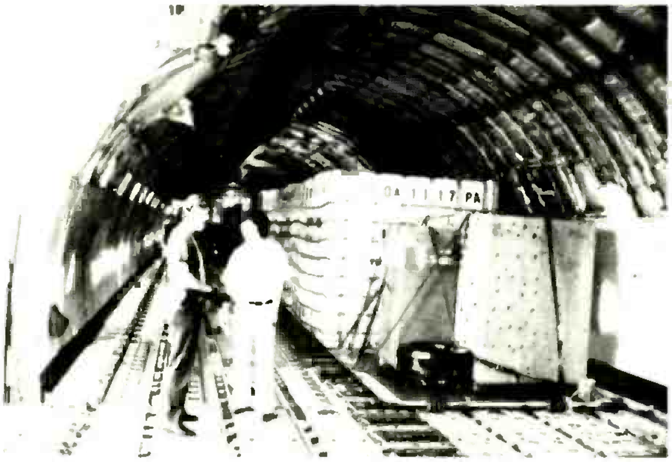
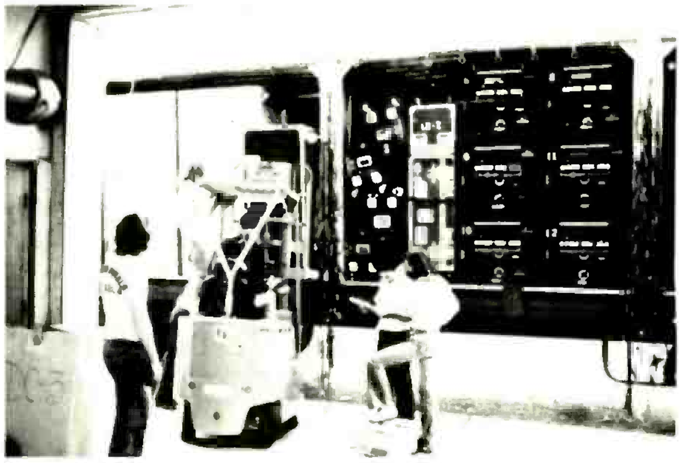


— *loading the M-1's . . .*



. . . aboard the 747 —

Shipping

Physically moving all this equipment to Brazil was quite a task in itself. Steve Oliker, of A-1, was responsible for most of the arrangements for consolidating and shipping the assembled equipment. Two 40-foot airline trailers, each containing a total of four M-1 containers had to be used. M-1 is a designation given by Pan Am to the largest airline cargo container in the world — which is 8 feet by 10 feet by 8 feet high. This container will only fit on a Boeing 747 outfitted for cargo exclusively — there isn't enough room for passengers when one of these is put aboard the plane (see Figure 2). Pan Am simply dropped the containers off at A-1's facility in Hollywood, and A-1 Audio personnel actually loaded the containers themselves using their own forklift (see Figure 3). Permanently marked on the warehouse floor at A-1 Audio are the sizes of all the various airline containers presently in use. Equipment is stacked up within these boundaries to get an idea of about how many containers will be needed for the shows they do. Additionally, the company has a full 4 foot by 4 foot loading dock-size scale so that everything can be accurately weighed before the containers are loaded.

The equipment was delivered to the airport four days before the plane was scheduled to depart. Not content to simply turn over the equipment to the airline and hope to see it all again in Brazil, Al took several steps to verify his containers' location all the way down the line. "I personally went down to LAX and went into the Pan Am 747 that our equipment was supposed to be on. I was at the airport prior to the loading, I watched our containers being loaded on the plane and then I crawled up into the airplane and personally verified the container numbers. It doesn't pay to just take someones word or rely on a handwritten sheet of paper. If the plane left and you didn't get a container on that plane, you're out of luck because there wasn't going to be another cargo plane capable of this kind of load for another week. Also, because of the scheduling of the planes, the containers had to be transferred onto a different 747 at an intermediate stop between Los Angeles and Brazil. We flew someone to that intermediate airport who supervised the transferring of the equipment from the first airplane to the second one. This was also arranged with the respective airport personnel ahead of time, because normally they don't allow anyone to go into a 747 freightliner. We feel these steps are necessary because a shipment of this size and importance does not happen by itself. What if someone were to go *home early and forget* that we had to have all these containers on a specific flight? There are no excuses because when we have to do a show, we have to do a show!" Are these precautions worth the effort? Anyone who has ever shipped equipment through O'Hare Airport, in Chicago, knows how easy it is to lose something, and it's usually the case with the microphones!

Once the equipment arrived safely in Rio and cleared Customs (another whole story), it had to be transported to the Stadium. Again, careful pre-planning was called for and extra time was allowed to overcome this obstacle. Obstacle? What's so difficult about moving 25 tons of sound gear through the streets of Rio? Well, the main thing that must be considered is that in many countries such as Brazil, all the transportation equipment that is commonplace in the United States is

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Frank Sinatra . . . 'Singin' in the Rain' For 140.000 at Rios' Maracana Stadium

just not available. Our domestic assumptions are just not valid, everything is different. Even if you think you know what you want, you may be in for a few surprises. For instance, when they say they will have trucks available, don't get visions of a fleet of enclosed Ryder 24-footers with lift gates and locking rear doors.

What Al and his crew saw on arrival at Rio were some 20-foot, open, stake-bed type trucks (decorated with little flowers and daisies painted on their sides.) These were all-purpose vehicles that couldn't be locked-up. Theft became a real possibility under the exposed circumstances. The trucks weren't covered so they had to consider the everpresent rain. They didn't have lift gates or ramps so they had to think about how to get their equipment on and off. In other words, they had to allow for a lot of extra time for these unusual circumstances.

Eventually all the gear got loaded onto four of these festively decorated vehicles which then made several trips back and forth to the Arena. At the aiport there was a single forklift available at the Pan Am Terminal. At the Coliseum they used "Brazilian Forklifts," which consisted of about 25 laborers and a few planks of wood. As it turns out, labor is cheaper than the gas for a forklift so that's the method that's used.

The Installation

As colorful and charming as this may or may not have appeared at the time, the crew was sure of one thing: the installation was going to take every minute of the seven days that they had available. In and out, the show had a maximum time limit of ten days during which they could abuse the grass playing field surface in the stadium. (And you just don't go about abusing the premier soccer field in South America) The show (as will be explained) was to be held on the eigth or ninth day. That meant that the stage had to be built and the sound system installed in under seven days. At best they had two days to take everything down. If they stayed on the field longer than ten days they would be blocking the sun and rain from reaching the grass for too long a period and the soccer field would be ruined. Destroying the biggest and best soccer field in Brazil is something to be avoided at all costs! Operational problems with the grass environment will be touched on later.

According to Al, doing shows South of the Border takes at least four to five times longer to set up than similar shows in the U.S. In Brazil the basic laborers are paid about $80.00 a month (there are no union regulations to deal with) so almost everything is a brute-force task, with little mechanical help. "In the morning we always had a full crew," Al remembered. "Then the siesta would come, which was a two to three hour break. It was so hot at mid-day that it was logical for everyone to take a break then. However, at the end of the siesta there were times when only half the crew would return. If you asked the crew foreman about this, he would just reply that 'Well, one goes out this exit, one goes out that exit, what can I do?'"

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Example of International Carnet Form used for no-duty customs clearance into foreign country and return to U.S.

The show was designed to be played in the round with two separate audience seating areas: one on the grass surrounding the stage for the big spenders, and the other in the normal stadium seating encircling the outside of the playing field. Instead of trying to cover both areas with a single sound system, Bob Kiernan asked A-1 Audio to provide a medium size inner system for the people close to the stage, and a high powered outer system located around the perimeter of the field for the grandstand seating. They used two digital time delays of approximately 100 and 150 milliseconds on the outer speakers so that all the sound would arrive at the grandstands at about the same time. The two increments of delay were necessary because the stadium is oval shaped. The distances between the outer and inner systems varied depending on which side of the field you were on.

In a stadium of this size you are naturally going to encounter a bit of reverberation, especially if the show is in the round. Even though it was open and outdoors, there was a tremendous echo that slapped right back to the center from the sides of coliseum. In effect this was largely overcome by the absorption factor of the 140,000 or so people and by increasing the overall level of the system. This was not a particularly difficult task, but it did take close to a full day to balance the levels and set the delays. There was so much area to cover that it took about an hour to physically walk around the perimeter of the arena.

Moats? Of course, there were the moats to contend with. Well, apparently Brazilian soccer matches are very exciting events and when you've got 150,000 very excited people in one place, moats come in real handy. At the edge of the playing field was the inner moat which was 10 feet deep by 8 feet across. Bordering this was a service road that was about 4 feet below the playing field, so someone would have to jump 8 feet across and 4 feet up to get across this first "moat." There was a second moat right in front of the stands that was about the same size. So all the equipment could only enter and leave the field from one side where a temporary bridge had been built across the inner moat. The musicians and Mr. Sinatra got to the stage through a tunnel which opened onto the field from underneath the stadium. There was one more special bridge built for the people who sat on the infield next to the stage. These people had paid about $100.00 a ticket so that they could actually see as well as hear Sinatra, and they weren't considered likely to riot.

The Inner Audio System

The stage itself was built in the shape of a six-pointed star which resembled a huge multi-colored Aztec sun laid out on the field. The orchestra was arranged around the

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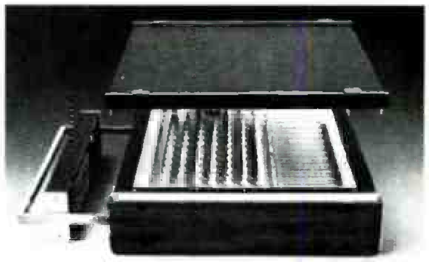

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center of the sun and the six rays, extending outward from the center, served as ramps that Mr. Sinatra could walk out onto during the show. In designing the set Bob Kiernan had arranged for the inner PA system to be incorporated into the stage itself so that the loudspeakers would be evenly distributed around the perimeter of the set. This inner system consisted of twelve bi-amplified monitor type loudspeakers that were three-way systems containing a 15-inch JBL or Gauss woofer, a JBL 2441 mid-range driver and a JBL 2402 tweeter. In addition there were eighteen small two-way speakers that had 5-inch cones and a small tweeter. These were actually mounted right in the facing of the stage and were for those people who were seated immediately next to the stage. The twelve bi-amp speakers were located back at the inner points of the star. There were no speakers over the stage because they didn't want to interfere with any sight lines or television camera angles. The stage was about five feet off the ground. The inner system was designed to reproduce medium power levels since its intended audience was in fairly close proximity to the stage. The idea was to deliberately not overpower these close-in people in an attempt to have the inner system affect the audience seated 300 feet away in the stands.

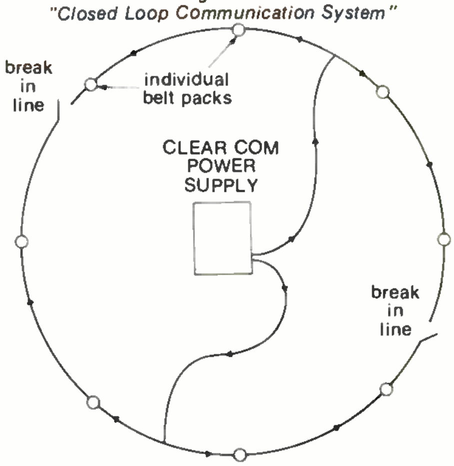
The Outer System

The audience in the stands was covered by a much higher power sound system which was located around the perimeter of the playing field and was fed through the two digital delay units. This perimeter system was made up of twelve VIP (Vertically Integrated Power) speaker stacks, so-called because of the manner in which they are assembled. These speakers were set up on the playing field side of the inner moat so that the sound had a chance to disperse across the distance between the two moats before it reached the people in the stands. Each individual VIP stack could therefore be run quite loud and could cover a fairly wide area without blowing anyone out of their seats, since the closest anyone could get to a speaker was at least 50 feet. Individiual VIP systems were then located about 90 feet apart all the way around the perimeter of the field. Given the 90° pattern of the horn and the distance to the seats, this spacing gave them about a 50% overlap in coverage from one seating area to the next. If any single speaker stack malfunctioned, that area would still be adequately covered by the two adjacent VIP systems. Of course, the level would be lower in this area, but at least the sound would still be clear and intelligible. This is an example of the importance A-1 Audio places on reliability. Everyone in the stands was listening to at least two speaker sections at all times. Each coverage section contained about 10,000 people!

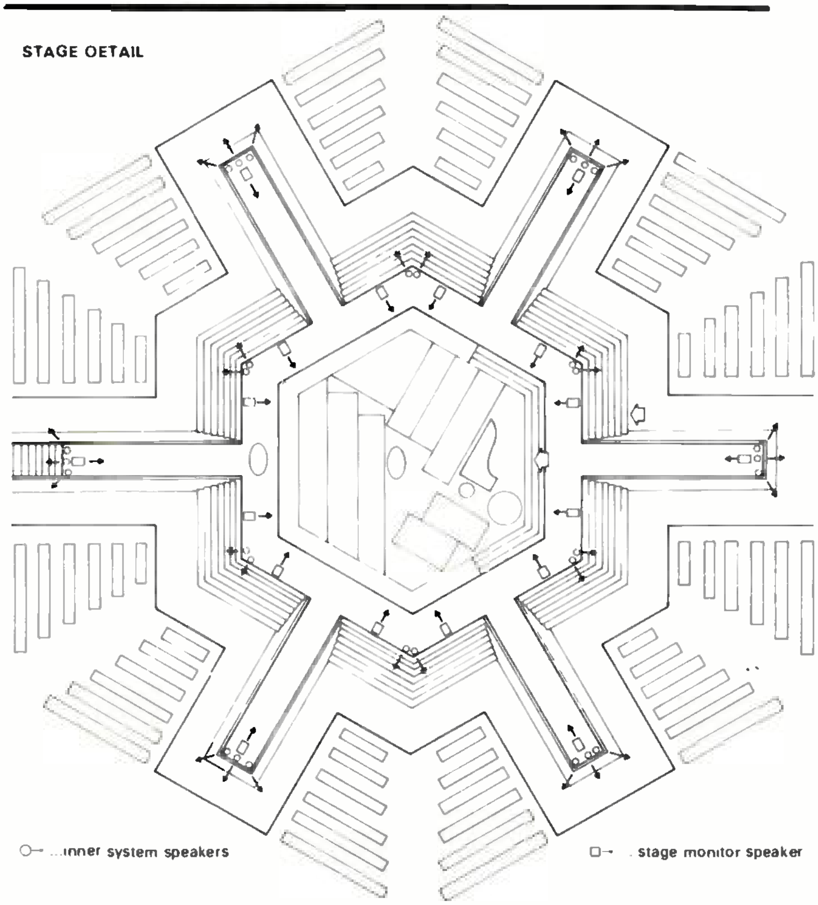
The location and layout of the speakers was all done from drawings and photographs of the coliseum before the staff ever left the States. Distances between the speakers, the stage, and the console areas were determined from scale drawings and the proper cable lengths were prepared. Everything was pre-labeled as to its specific use so that time would not be wasted on location. It took quite a long time just to lay the cable out, mainly due to the fact that distributing the speaker system around the perimeter of a soccer field uses up a heck of a lot more cable and installation time than if the show were being done from one end of the field, as is normally the case. A-1 Audio was also responsible for installing the 45-position intercom system which included running cable along the top outside rim of the stadium to the follow-spot lighting positions. The ellipse of the Maracana Stadium was about 1,000 feet across at its widest point, and the perimeter around the top where the follow-spots were located was about 3,000 feet in length. The top of the grandstand is about 100 feet, or about 10 stories high. You can now get an idea of how much intercom cable they had to use. Since it took about 3,000 feet to get around the perimeter and another thousand feet to get down on either side, it ended up being about a mile run altogether. The intercom system used by A-1 is made by Clear-Com and was hooked up in a novel manner which again underscores A-1's concern with system reliability.

They ran what Al refers to as a "closed loop communication system." First, they ran the Clear-Com cable in a completely closed loop to all the spotlight stations. Then they picked the two furthest extremes of the loop and ran two separate lines down to the Clear-Com main station which was located at the console. A break could occur in two places along the loop and complete communiction would still be maintained (see Figure 4). There are a lot of connections in a mile's worth of 50-foot cables, so they figured it would be reasonable to assume that there might be a possibility of failure at an inopportune moment and therefore designed the system so that there could be two failures in the line and the show would go on as planned. There was so much space to cover and so much wire to trace that it was quite possible a fault couldn't be corrected during the time span of the show — hence, the closed loop. The Clear-Com is so wired that you can come out of an output at the main station, loop through a series of belt packs, and come back into another output on the main station using a three-pin reversing connector. You could actually come out of the main station with a third cable to the same multiple belt pack loop for an additional back-up line if the situation warranted it.

Figure 4: "Closed Loop Communication System"



○ ... 51 belt pack stations for sound and lighting, around three quarters of a mile of wire.



— continued overleaf . . .

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Frank Sinatra . . . 'Singin' in the Rain'

The VIP System

The VIP is a three-way tri-amplified system in which the bass units reproduce the low end response from about 25 Hertz to 500 Hertz. The mid-range section consists of two precisely coupled horns which operate from 500 Hertz to 5,000 Hertz — at which point the tweeters take over. There are eight tweeters in each cabinet which are set in a 90° convex curve that focuses back to the same vertical axis that the mid-range drivers pass through. This point is a little further back from the center of the woofers' voice coils. According to Siniscal this physical alignment of driver elements was developed using the Hewlett-Packard 3580A Spectrum Analyzer and achieves approximately the same result as that obtained from the latest electronic delay techniques. The bass cabinet (Figure 5A) contains two 15-inch Gauss 4583 woofers rated at 400 watts RMS apiece in a bass enclosure which measures 7 feet tall by about 3 feet wide by 4 feet deep. Each 8 ohm woofer is driven by one-half of a BGW 750B stereo amplifier which is mounted in the bottom of the cabinet itself. This assures a very high damping factor and excellent direct coupling to the transducers.

There are two AC connection points to the amplifier mounted on each cabinet . . . either one of which will power the amp (see Figure 5B). Whichever one is not being used is automatically shut off by a relay which is in turn powered by the AC from the connector that is being used. The AC sockets are located at the top and bottom of the cabinet to simplify hook-up — depending on whether they are being wired from above as is normal in a hanging situation, or from below. This makes for a cleaner cabling set-up. After all these bass cabinets are seven feet tall!

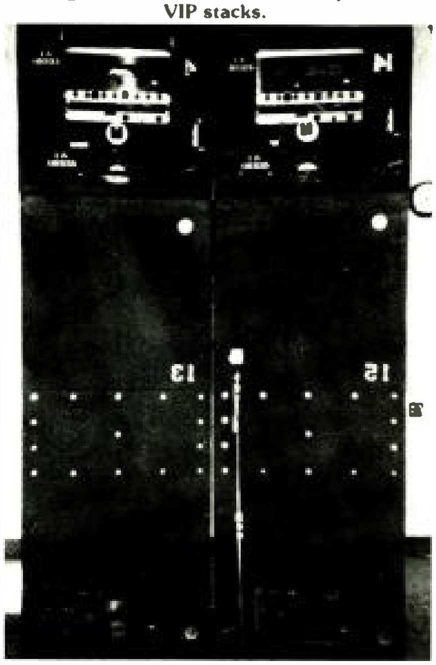
Figure 5A: **Two VIP Series stacks with mid/high cabinets mounted on top.**

According to Al, "the bass amplifier has a tremendous amount of reserve capability and since it is running at 8 ohms and not at 4 ohms, the amp runs much cooler. Also, the reserve power that is available enables it to cleanly reproduce a signal that contains a lot of repetitive low frequency sounds. The amp is rated at about 250 watts into 8 ohms, so the speaker can withstand about twice as much level as the amplifier is capable of putting out without clipping."

The mid-range/high frequency cabinet (Figure 6) houses two JBL 2482 Compression Drivers attached to two JBL 2350 horns. They are very carefully aligned in the vertical plane at 4,000 Hertz in order to get them to couple efficiently. Al states that, "The advantage of doing this derives from the fact that two drivers coupled are theoretically equivalent to the output of four drivers. In reality, however, it ends up being equal to about 3½ drivers because things don't couple quite that perfectly even though we align them as best we can. By permanently aligning the two horns, we have more of a unified mouth area than you can get by simply stacking two separate horn cabinets on top of each other. We also achieve a lower cut-off frequency than is possible with a single horn, due to the fact that the larger area provides for a smoother transition to the low frequency." All the internal surfaces of the horn enclosure are packed and sealed with a high density pressurized foam which is poured into the cabinet once the horns are aligned in place. This foam is originally a liquid which expands and hardens in about fifteen minutes to permanently lock the two horns in place. This foam also effectively damps out any resonances inherent in the horns themselves and prevents the ringing that is largely responsible for the typically nasal sounds of many horn loaded mid-range cabinets. The result, says Al, is a much more natural sound which is especially noticeable on vocals.

The power connectors to the high frequency cabinet are of the twist-lock type

Figure 5B: **Rear view of two complete VIP stacks.**



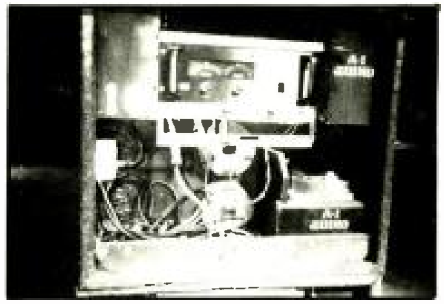


Figure 6: **Rear view of VIP mid/high cabinet.**

to ensure that power cables don't vibrate or otherwise pull loose during a show. Again, they use a single BGW 750B amplifier in each cabinet to power the mid-range and the high end. The two 16-ohm JBL 2482 mid-range drivers are connected in parallel — thus placing an 8-ohm load on only one channel of the amp. This configuration results in at least 250 watts being available to power both of the drivers which effectively prevents the amps from going into clipping. A-1's design philosophy is to eliminate limiting at the amp in order to preserve as much dynamic range in the system as possible. Instead they prefer to utilize speakers that can withstand more signal level than the amps are capable of producing before the onset of clipping. The other half of the 750B amplifier powers the eight JBL 2402 tweeters which are wired in a series-parallel configuration. "The reserve power capability for this combination of components is absolutely fantastic," Al states. "The amplifier is just coasting all the time." As with the bass cabinet, the 750B amp is integrated into the back of the high range cabinet.

The line level audio signals are brought out to the power amps from the console area via an 11-pair snake cable. They can also be transmitted via three independent cables utilizing standard three-pin XLR connectors to carry the separate low, mid, and high signals — although the snake is generally preferred. This 11-pair cable uses a military type jam-nut connector which provides a very quick positive connection and eliminates the possibility of stripped threads. The 11-pair cable provides the flexibility of sending three separate three-way signals down the line, i.e., left low, mid, high; center low, mid, high; and right low, mid, high — as well as a single full range signal. A three-position switch on the back of each VIP system then selects which one of the three three-way signals it will amplify. The concept behind this came from Grey Ingram who needed a system for The Doobie Brothers that would allow him to run a common cable to all the cabinets and still maintain flexibility in the hook-up. Each cabinet could then be switched over to left, center, or right depending on its position in the system. Even though the system was run in mono in Brazil, these three channels proved to be very useful indeed. This feature allowed both delayed signals to be available at each VIP stack at all times without having to switch between feeds at the console. Initially, it was difficult to determine which delay would be most appropriate for each speaker stack since they were all located on an ellipse relative to center stage. Having the ability to switch

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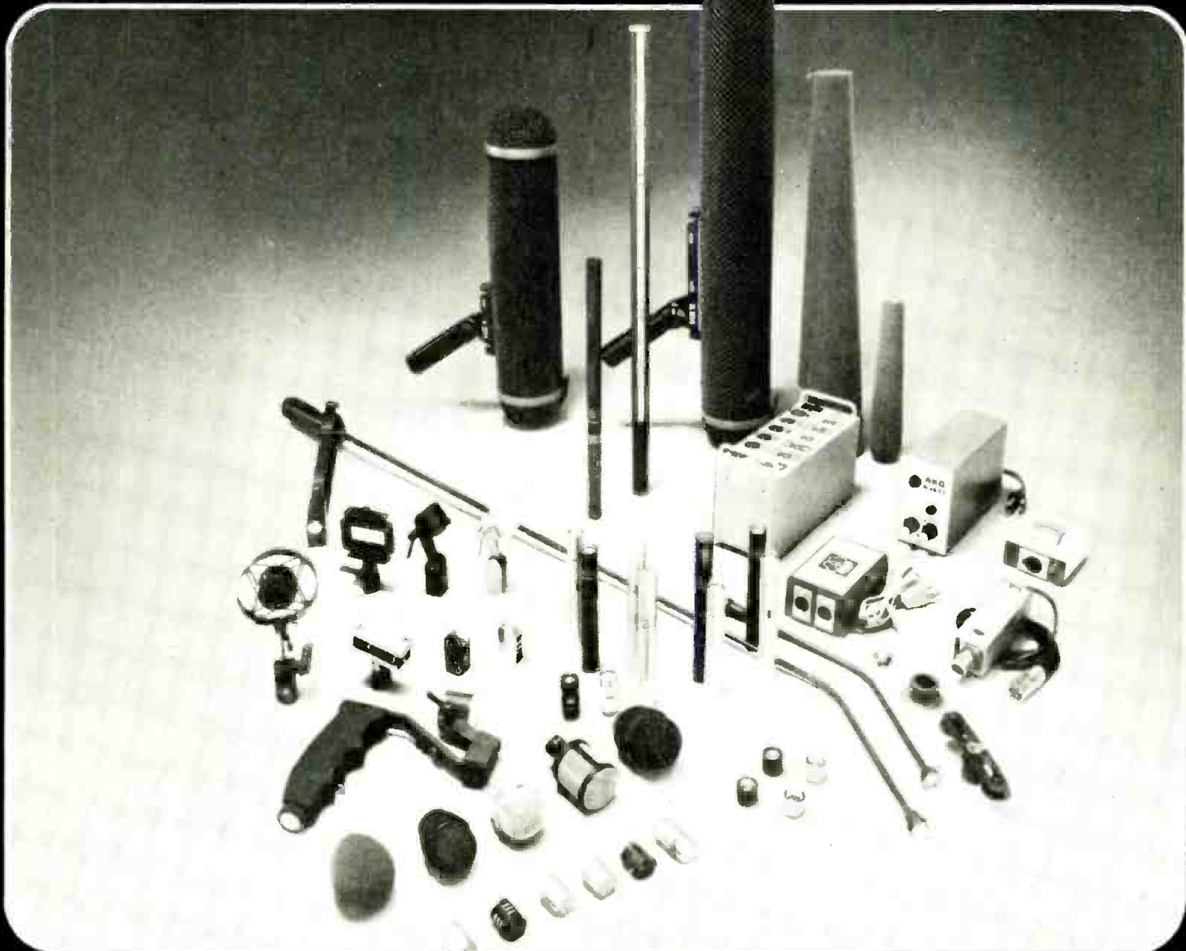

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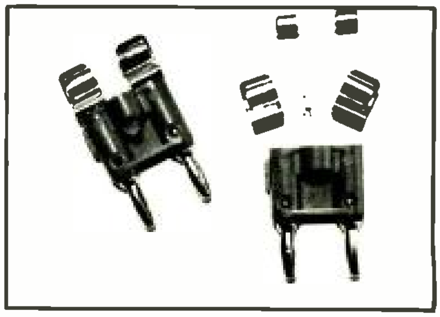
from one delay to the other at each individual speaker made this particular task quite easy and saved a considerable amount of time and cabling. "We simply had a guy with a walkie-talkie go around to each VIP speaker and switch from one delay to the other while another member of the crew listened in the stands and communicated back which delay sounded best," Al recollected. "This had a lot to do with getting a good quality sound in the arena."

And now we come to another clever idea that scored a 7, a 7, an 8, and a 9 from our hard-nosed judges in the *American Ingenuity Finals*. Located on the back of each speaker cabinet and connected between the output of the amplifier and the input to the speakers is a clever banana plug connector which has several uses. First of all, the bass cabinet is quite rigidly built and solid and it would be extremely difficult to get at the terminals of the loudspeakers for testing. Then, too, being able to quickly and easily open bass cabinets tends to undermine their rigidity in the first place. So, instead, A-1 Audio brings out to a test point both the output of the power amplifier and the input to the speaker on a standard banana plug socket. Dual banana plugs are then used as jumpers between both the plus side and the minus line to each speaker — with the minus jumper being normally hard wired across the two prongs. As an added safety factor, however, the plus jumper is not simply hard wired. Instead A-1 Audio has installed a standard fuse holder to the plug through the little holes where the set screws normally go. A fast acting, low resistance fuse was then selected to prevent damaging signals from reaching the speakers. Then, wired in parallel with this fuse, there is a tiny, 48-volt incandescent light bulb that fits right into the round wire opening on the side of the standard MDP banana plug. So in the event a fuse ever blows, the resultant open circuit forces the signal to flow through the light bulb, causing the bulb to light and thus indicating which cabinet has malfunctioned. Since the light bulb has a resistance of its own, the amount of signal reaching the speaker is lessened considerably and damage prevented.

Now, obviously, a normal off-the-shelf fuse is not the most precise device to begin with and can blow at anywhere from 200 to 400 per cent of its value. For this reason A-1 uses only precision made 4-amp fuses that blow a lot closer to their rated value. The company puts a great deal of emphasis on operating the system correctly in the first place so that the fuses are never called to duty. In the unfortunate event that something rude *does* happen, it's a lot simpler, faster, and considerably cheaper to replace fuses than loudspeakers. Especially if the fuses are already attached to banana plugs that can be quickly inserted. These handy little plugs are available directly from A-1 Audio, pre-wired and fused, for $19.95 each.

As was pointed out earlier, the VIP system is designed with the power amplifier mounted in the speaker enclosures themselves. Al describes his reasoning behind this approach: "The purpose of this is first to improve the damping factor of the system by essentially eliminating the speaker cable. Trying to transmit high speaker level signals long distances and dealing with the problems of large resistances through the wires didn't make as much sense to us as did using normal mike cable to carry line level signals. This, of course, was a big help in Brazil where individual cable runs to the VIP systems on the perimeter of the field would have been incredibly long. Also, the possibility of phase reversals is eliminated since the cabinet is correctly wired when it is first assembled and then it is left alone." But what about a normal concert situation wherein they have a stack of speakers on each side of the stage and an amp goes out in a bass cabinet, for example? A short 3-foot cable with banana plugs at both ends is all that is needed to parallel patch into a neighboring amplifier. Of course, this amp is now looking at 4 ohms instead of 8 — which actually is not a problem at all since the BGW 750B is rated at this load to begin with.

. . . winner in the 'American Ingenuity Finals' —



This author's main concern with this approach had to do with the system's recoverability from a malfunctioning amplifier in a concert situation in which the system is hanging at least twenty feet in the air — a bit out of reach for most of us. "If a problem developes during the day, at sound check for instance, we will change out the amplifier entirely," Al explained. "But if it happens during the show, we probably won't do anything unless it is in a critical location — at the end of the cluster, for instance — where its failure would be very noticable since that area isn't overlapped by another system. We usually have so many speaker systems up in the air that it's not a serious problem if one goes out. If it's critical enough, however, a rigger will go up and make the necessary patch. Generally speaking, once it is operating and in the air, it is extremely rare that anything goes wrong. If a problem occurs at all, it is usually due to the handling the unit gets when it is loaded, driven 500 miles, and unloaded again night-after-night. Also, the systems are hooked up and tested while they are still on the ground. We feel that the advantages of having the amps located in the speaker enclosures — even though they are hung twenty feet above the stage — far outweigh any disadvantages that might be encountered. The chance of it failing in the air — well, it just doesn't happen. One of the secrets, of course, is to use reliable amplifiers."

In addition to the BGW 750B, A-1 Audio uses quite a number of Yamaha P2200 amplifiers as well. All the systems that were taken to Brazil, however, utilized the BGWs due to their quick-change heat-sink assembly which contains the 10 output devices and all the input circuitry. Once you remove the top of the amplifier, you can unplug the whole channel from an octal socket and plug in a new one in a matter of moments. They took several spare heat-sinks to Brazil, but were fortunate in that none of them had to be used.

The Monitor System

The monitor speakers were composed of a 15-inch JBL K130 or Gauss 5840 woofer; either a JBL 2441 or 2440 compression driver for the mid-range; and a JBL 2402 tweeter. Each of these three-way monitors was bi-amplified by a single BGW 750B amp with one side of the amp powering the woofer and the other side driving a passive crossover that fed the mid-range and tweeter elements. Eighteen monitor speakers were used in Brazil for Mr. Sinatra, and were actually inset into the floor of the stage itself. One was located at the end of each of the six sun rays and there were two in between each ray around the center part of the stage which accounted for the remaining twelve. These monitors were all fed from a central point individually so that any one line could malfunction and the rest of the monitors would not be affected. Al said he went to this extreme because, "After all, Sinatra had to hear in order to do the show. There is no 'Take Two!' I think it is extremely important in sound reinforcement to realize that there is no 'Take Two.' There is only 'Take One' and it's not 'Take One an hour from now,' it's 'Take One right now!' This is one area that I spend probably my whole life working on and my company is geared toward maintaining this attitude."

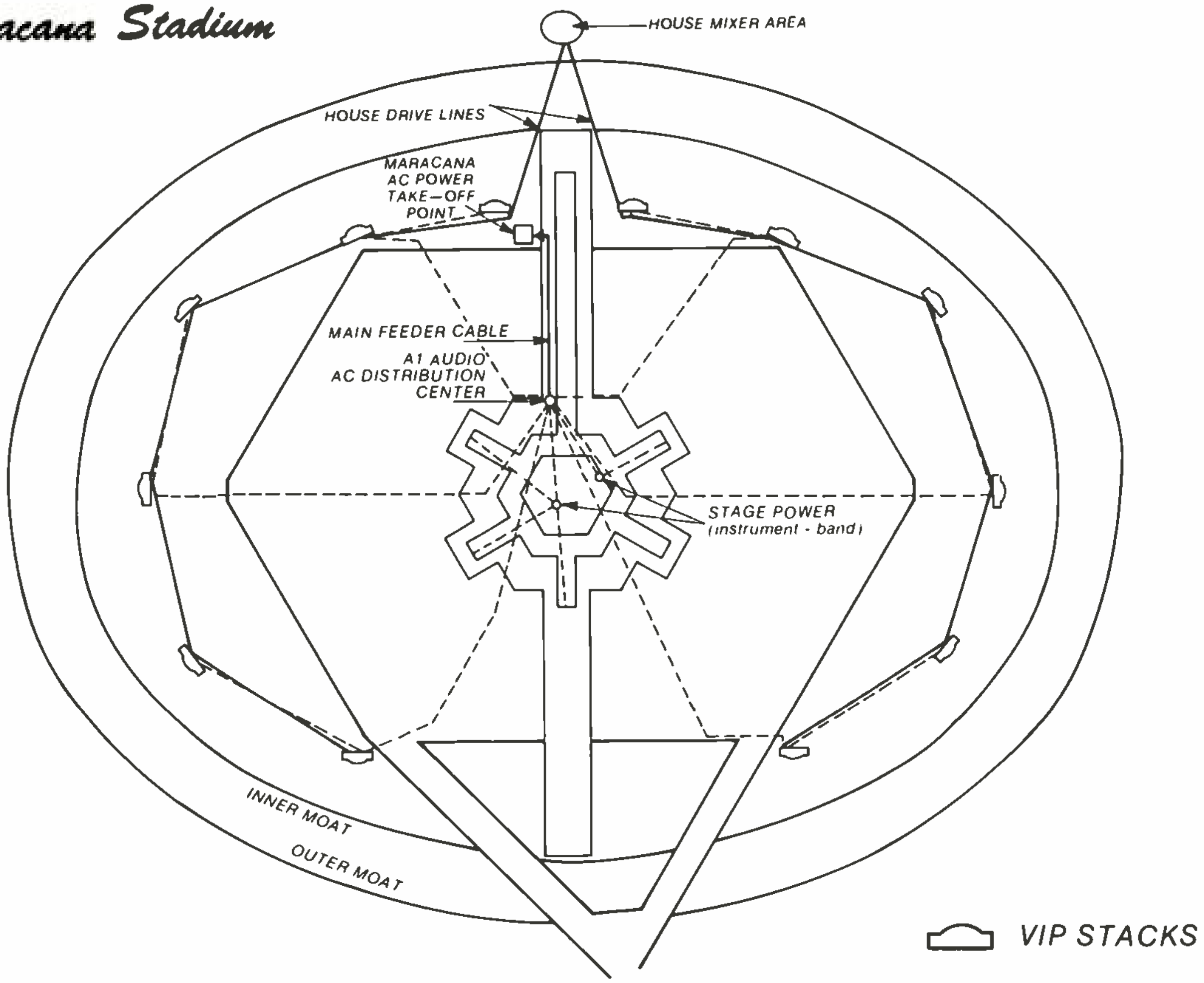
Crossovers

The crossovers for the VIP system were Yamaha F1030s that were located at the console position. These are three-way crossovers with separate controls for the low, mid, and high frequency turnover points, and which contained a visual clipping indicator. The frequencies are front panel selectable and can be rolled off at either 12 or 18 dB per octave, on either side of the crossover points. This flexibility allows you to play some games with the normal response of your system. For instance: say you were in an environment where you wanted a little more build-up between 500 Hertz and 800 Hertz. You could therefore set up the woofer to go out to 800 Hertz with a 12 dB per octave roll-off and then extend the horn down to 500 Hertz with an 18 dB per octave cut-off. Now, I'm not saying that this is the best way to achieve this result, but only that it is a possibility with this crossover. All the various settings on this Yamaha crossover are absolutely repeatable because all the controls are detented. "I worked with other manufacturer's crossovers where they talked about how nice it was to be completely variable," says Al. "To me that is just absolutely impossible to work with. It is unrealistic to believe that when the unit bounces in a truck for 500 miles, it isn't going to shift a little — or you accidentally brush against it and it changes. Well, I just can't tolerate that much shift. Another nice thing about the Yamaha F1030 is that there is a phase reversal switch right on the back of the unit. You can throw the mid-range in or out of phase with relationship to the high end just by throwing a switch. It's a fine crossover and works very well for us."

A somewhat modified JBL 5234 stereo crossover with changeable frequency cards was used for the monitor speakers. The first modification involved installing Jensen input transformers in place of the unit's differential input to guard against damage from stray AC on the line and other dangers peculiar to road work. A custom designed 18 dB per octave, 20 Hertz, low-cut filter was installed in the crossover to limit the low frequency response. According to Al, there isn't anything coming out below 20 Hertz that's useable anyway — just a lot of rumble, stage noise, and the impact from the occasional dropped microphone.

Sinatra's monitors were run off the house console which was a Yamaha PM2000, 32-in by 8 outboard. There was no separate stage monitor mix as they felt it was unnecessary with only one vocalist and they really didn't need another level of complication added to the show. Keeping everything as simple as possible, yet flexible enough to do the job, is a key point in maintaining system reliability. A-1's dedication to reliability is evident in all phases of their operation including the

Maracana Stadium



construction of the road cases. All their equipment racks are of the "case within a case" design. They first build a very rigid inner frame out of 13-ply Baltic Birch Plywood on which they mount front and back rack rails. This enclosure is then placed inside a foam-walled shipping case which is made with removable front and back lids. Any shock given to this case is absorbed by the foam and the force to the rack is evenly distributed and significantly reduced. They preferred a front and back lid approach as opposed to the lift-off top concept which can result in a lot of large empty boxes stacked on stage and in the way.

Cable reels are used for storing and transporting the microphone cables. All mike cable are labeled with different colored strips of heat-shrink tubing according to their length and there is a sample piece of color coded cable on top of each reel. Stagehands are then told to simply connect all similar color coded cables together and wind them onto the reel with the matching sample strip. This also provides for a fairly good way of checking the cables out ahead of time since the end cable comes out of the hub of the reel and a cable checker can be used between the first and last connector to make sure that all the cables on the reel are okay. The main advantage to this system, however, is the fact that the cables are not all knotted up. Extensive use of sub-snakes also cuts down on the length and number of individual mike cables needed.

Microphones

AKG C451 condenser microphones were used extensively on this show, especially on the large twenty-piece string section. Contact pick-ups were avoided in order to get a more natural ensemble-type string sound. Sennheiser 421s, Shure SM-57s, and SM-58s rounded out the microphone complement. A Helpinstill piano pick-up was used in conjunction with some live overhead mikes to pick up the piano. Direct boxes equipped with Jensen transformers were used and are manufactured and distributed by A-1 Audio themselves. AKG mike stands were used on this show and are favored by A-1 due to their robust nature and collapsible bases which simplified shipping considerably Atlas type stands with screw-on bases are also available if requested by vocalists.

Microphone Splitter System

The splitter system was transformer-isolated with direct lines feeding A-1's equipment and an isolated split going out to the television truck. The snake connectors are circular military multi-pin connectors that are ordered from a military supplier and come with locking jam-nuts to again avoid the crimping and stripping problems common to screw-on connectors. The mating pins are all gold-flashed and are quite durable. A-1 uses the AMP connectors as well, but chose this military approved system for Brazil since it has a very tight seal that is superior in keeping out moisture — one of the major factors they had to contend with. In this particular case the transformers in the splitter boxes were Triad A66J, which are actually console input transformers. Jensen transformers, however, are used in all the splitters currently being built at A-1 Audio. This splitter provided for 36-input and 10 return lines, while two additional 6-pair snakes carried signals to various speaker systems. So there were about 60 lines in all carrying audio information between the console and the stage. The isolated split that was made available to the television company terminated in three-pin XLR male connectors which they then plugged into their own input snake box. The audio connectors and wiring configuration of their snake was the same as A-1's, so there was no problem at this point. But, as you may have guessed, a problem did crop up. Personally I don't trust frequencies that I can't hear and I always approach television tie-ins with one eyebrow raised. A hum problem arose due to the fact that the television truck was parked quite a distance away and was tied into a different AC power source than the one used by A-1. A potential difference was created that was aggravated by the long feed-line and wasn't eliminated until the television crew grasped the fact that they had to be tied to A-1's AC system. For some reason they didn't feel it was at all necessary and this turned out to be one of the biggest problems Al and his crew encountered. This is, no doubt, where

Pat Weber's technical interpreting expertise earned him his T-shirt.

AC Power

The power that was available from the Maracana Stadium was three-phase at 100 amps per leg, with a neutral and no ground. A-1 used two of the phases and the neutral and drove a copper stake about four feet into the ground at their power distriubtion point to provide a solid gound just as a safety measure. They could have used a floating system, but felt it was better to provide their own ground especially since there was so much rain at the time and the grass was always soaking wet. By the way, this is the first time that a concert had ever been held in this coliseum, so they had to be extra careful that it was safe and accident-free. They brought their own power distribution system and feeder lines and tied right on to the main power source located under the grandstands. The power distribution alone was quite some project as thousands of feet of AC cable were required just to power the outer VIP systems.

"We always trace the power back to its source," Al maintains. "I feel it is important that we walk the line back and see where it's coming from. We made certain in Brazil that we were isolated from the lighting system, for instance. We also brought our own electronic instantaneous voltage regulator to provide a stable source of AC for the audio consoles and all the low level, signal processing electronics, reverb chambers, etc. In foreign countries you can never be sure how stable or reliable the power is going to be. There may be some instantaneous shifts or power spikes that come down the line. The voltage regulator that we used to protect us from these fluctuations is capable of handling about 30 amps and puts out exactly 117 volts AC if it receives anything from about 95 to 135 volts. It can be set to produce 120 volts AC or 115 volts AC or whatever else you want. Anytime the incoming power drops below 95 volts or rises about 135 volts it changes the output on an approximately linear scale. In other words, if you have 140 volts AC across the input to the regulator, it's going to put out 117 volts plus 5 volts for a total of 122 volts — at which point the internal regulators of the power supplies in the console start working. We have found that this is a mandatory system to use when traveling in foreign countries — especially in a country that is not highly industrialized." There was no regulation on the AC lines that fed the power amplifiers, but since they were not drawing a tremendous amount of current, there was plenty of leeway in their operation. In Brazil, because of the tremendous size of the event, extra power was brought in which helped to keep the power stable. The changes in the line voltage were fairly small with the input to the regulator ranging from 126 volts to about 118 volts AC. "But we have noticed great changes in Mexico," Al stated. "We had a situation once in Mexico City where the power amplifier in a small speaker/amp was not plugged into a regulator — it caught fire and the speaker burst into flames on the stage."

An RCA Line Voltage Meter was connected across the output of the voltage regulator to make sure that it was operating correctly and producing a steady 117 volts. A-1's power distribution system has a digital voltmeter and a digital ammeter connected between each phase and neutral on the primary side of the panel between the master fuse and the bull switch. This enables them to make certain that the power is correct before throwing the switch that feeds their equipment at 117 volts. A 220 amp breaker powers several sub-breakers and it is to one of these sub-breakers to which the instantaneous electronic voltage regulator is connected.

Protection From Rain

Unstable power was not, as it turned out, a problem in Brazil. The real problen was rain — and lots of it. The consoles, microphones, cables, power distribution, speakers — in short, everything — were subjected to a downpour every single day. In order to ensure that it would all still work when the downbeat came, Al and his staff took several precautions that are generally unnecessary in a normal concert situation. They used Switchcraft Gold-flashed XLR Connectors on all mike cables and suspended the cables in the air whenever possible. They used only high-quality Beldon 8412 and 8413 mike cable (which was made especially for them with their name imprinted every foot or so right on the cable). All the connections along an audio cable run were put up on pieces of wood. If

continued overleaf . . .

they were in a highly wet situation, tape was wound around the connection. "The signal shield in a microphone cable is *not* tied to the case of the connector," stated Al. "That mode of wiring could have caused terrible problems in the watery situation we had down there." Critical connections were covered up with plastic and the ends taped to make them waterproof. Several rolls of heavy-duty plastic were used to cover everything on stage as well as the power distribution, the audio consoles, the processing rack, and everything else exposed to the open air (see Figure 7). Now, since it was actually raining during rehearsals, a way to protect their expensive microphones without physically removing them and stopping the rehearsal had to be devised. How do you set up a $350.00 microphone in the rain anyway? Most people don't even want to get a speck of dust on them — much less have the diaphragm and electronic components subject to that much moisture. The solution? "Baggies" from a Brazilian supermarket. The same kind you would pack your cheese, avocado, and sprout sandwich in for lunch. Acquiring the thinnest Baggies they could find, they were put over the microphones and secured with rubber bands. Sound checks were done with th Baggies on all the time. Very thin polyethelene, they found, did not significantly reduce the transmission of sound. The Baggie wasn't stretched tight over the mike at all — in fact it was quite loose. Having realized it was going to rain, the crew had brought down a much thicker plastic with which they had originally intended to cover the mikes. But this turned out to be too thick to keep on the mikes all the time. The particular gauge of plastic that works the best is the kind from the supermarket used to bag your own vegetables — if you can ever get them open. Strong, hefty, expensive Baggies don't work as well as the thinnest, cheapest ones.

I must admit to a bit of skepticism of this whole affair — especially regarding the fact that something cheaper might actually be better — so I experimented with four or five different gauges of bags and Baggies myself and found that the vegetable bag is indeed the best. It's also rather humorous to see someone singing into a brightly stencilled carrot! On the basis of one grape per bag you can get a whole orchestra worth of protection for under fifty cents and have a healthy snack at the same time. Of course, there is a slight quality difference in the sound from the microphones but it really isn't very much! There is no need to punch little holes in the end of the bag either — acoustcially the Baggie just doesn't seem to be there.

"Subjectively it was outstanding," Al told me. "It really had a very minimal effect on the sound.

"I was recently talking with Ron Means, the professional division manager at JBL, about how to best protect the mid-range drivers in a highly wet environment. Even though the driver comes with a protective screen further back in the throat, he suggested putting a piece of polyethelene — normal Baggie material — between the horn and the opening of the throat at the end of the driver. In all the drivers that went to Brazil that January, however, we installed an additional protective screen of fine metal wire mesh right where the horn bolts onto the surface of the compression driver. We called both Gauss and JBL and got the same info from both of them regarding the appropriate size of screen to use. I also talked with Algis Renkus, formerly with Emilar and now with Renkus/Heinz, Incorporated, and discussed the problem of putting an additional layer there. This was done before anything was shipped down to Brazil because we wanted to have a second layer of defense keeping the moisture, the condensation, and the dirt out of the drivers. When showtime comes we can't have water in the phasing plugs!"

— rain . . . the afternoon before the show . . . baggies . . . more rain . . .

Another step that was taken to guard against the rain was to cut up pieces of cloth which were then kept in the throats of the speakers at all times except when actually doing a sound check. The entire speaker stack was then covered with heavy plastic which was tied down with rope to prevent the wind from blowing it off during the night. These precautions can apply anywhere where there are outside shows — i.e., in the outdoor summer theaters in the Midwest and on the East Coast where summer storms are not at all uncommon. The additional protective screens were not taken out of the drivers when the equipment returned from Brazil. In fact, Al liked the idea so much that he is incorporating it into everything he has. Some of these drivers were disassembled to see if there was any damage or corrosion and they proved to be just fine. Some of the screens were dirty, of course; but were easily cleaned since they were located right at the throat of the driver. This eliminated the risk of possible damage to the driver from an air hose or contamination from normal handling.

The Show

All these protective measures and weeks of careful pre-planning were finally put to the test the day of the show. As Al remembers, "It rained the afternoon of the show but that didn't stop the people from pouring into the stadium as fast as the rain did. They parked their cars in the middle of the streets around the stadium and the whole area was soon blocked to traffic. It was a big event in Brazil, and since it was being televised live all over South America, the show had to start by 9 p.m. at the latest. The show was scheduled to start at 8 o'clock, but they would keep the stations on the air until 9 o'clock if there was a delay due to rain. If the show didn't start by 9 o'clock, they would shut everything down, and in typical South American fashion, they would come back do the show the next night. It is almost impossible for us to believe that they were going to tell 140,000 people to go home and come back tomorrow, but nevertheless that was really the plan. So at 7 o'clock the night of the show it was raining. Nevertheless there were 140,000 people in those stands and everyone was cheering — the people on the field in the expensive seats were getting very wet and nobody seemed to mind. There was no pre-show entertainment; no lead-on acts at all. There was just the anticipation of Sinatra. It wasn't really cold since it was summertime, but it was definitely wet!

"You can just imagine the possibilities for failure. It's raining. It's just prior to showtime and we've still got all of our speakers covered up. We've got cloth in all the throats. We've got plastic Baggies over all the mikes. We've got all the power distribution covered up and we've got the consoles covered up. Imagine trying to do a sound check under those conditions.

"At 8 o'clock when the show is scheduled to start there is still a heavy rain falling. The spotlights are starting to fire up and you can see the lights shining down into the arena. Thirty spotlights cutting through the falling rain is a very dramatic sight! That, together with the roar of 140,000 people is just phenomenal. But even though it's 8 o'clock you can't very well uncover the piano, and the string players — especially Sinatra's string players with their Stradivarius violins — are not about to come out and play in the rain! At least they certainly don't want to. Needless to say, the television people don't want water on their expensive camera lenses either. The show has got to start pretty soon or everyone has to come back tomorrow night — a situation nobody seemed to mind too much, except for me, that is. I was sort of in a panic! To feel that we would not be able to do the show was such a tremendous letdown. The excitement was just tremendous! The musicians finally said that as long as it wasn't pouring down rain, they would go out there and play and would just keep drying off their instruments. Sinatra's people were extremely willing as well. Everybody was extremely willing — but it was just a matter of not being able to do a concert in a downpour.

"The local Brazilian people were used to this and took it all for granted. Nobody was concerned among the Brazilian people. If they had to come back tomorrow night, they would all come back tomorrow night. No big deal! Well, it's now about 8:30 p.m. and it's still raining. But it slowed down a little and by 8:45 it was down to a light drizzle.

"Then at 8:53 the rain had essentially stopped and the word was given to start the show. So we had only seven minutes to start the show! We had laborers placed at all the strategic positions on the field whom we had trained during the week to prepare for just this kind of situation. After all, there were only three of us on this job plus, of course, Bob Kiernan on whose capable shoulders fell the responsibility of giving the technical go-ahead for the show. I must say that Bob did an outstanding job. He was just superb. So as soon as he gave us the word we uncovered everything, took off all the Baggies, got the system completely operational in six minutes and started the show at one minute to nine."

— 150,000 seat Maracana Stadium . . . small arena at top is same size as L.A.'s Forum,

"MY GOD!"

"Mr. Sinatra walked out of the underground tunnel, came out onto the stage, and when he saw 140,000 people surrounding him, said just two words . . . "MY GOD!" He then walked up onto the center stage and did a great show which which lasted just one-hour-and-fifteen-minutes, including numerous standing ovations. Now it must be remembered there were no warm-up acts, no comedians, no Mariachis, no closing bands, no anything — it was just Sinatra. The show started at 9 o'clock and it was over at 10:15, and it was just fantastic! The lighting people who were up on the rim of the stadium told us later that the sound was absolutely perfect. We got tremendous reviews on the quality of the sound because it was evidently the first time anyone had ever been able hear anything correctly in that stadium. Smaller events had been attempted before, but either the equipment wasn't set up properly or they didn't allow enough time, or they didn't have an inner and outer system. The Brazilian press had even been a little apprehensive going into the show. All down the line they were very concerned about whether the people would even be able to hear the concert at all! But, as the lighting people and the engineers from the television stations said later, the sound was absolutely crystal clear. It was free of a lot of reverberation effect; was very intelligible, and the string sound especially was just beautiful."

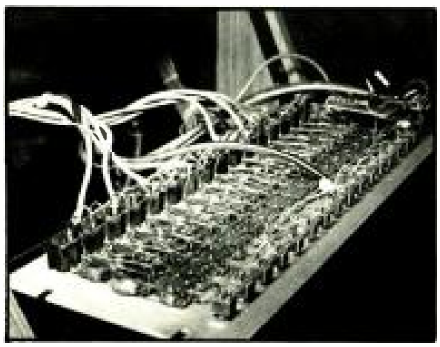
Then began the long arduous load-out. Because of the slow labor situation down there, it took all of that night and the next two days to get the equipment to the airport. After all, they again had to deal with that marvel of human engineering, the "Brazilian Forklift" — twenty guys and a couple of pieces of wood! □□□

Expand Your Console With An Outboard Solo/Cue System

by
Ethan Winer
photography by
Pete Hodgson

Many times during a recording or mix-down session it is desireable to be able to isolate or "solo" one instrument or track without having to disturb the rest of the mix. While nearly all professional recording consoles being sold today do have some type of solo switch associated with each input, many of them do not offer what is known as "solo-in-place." A solo-in-place system does exactly what you would think a solo system should do — that is, it mutes or turns off all of the channels except those being soloed. When used in the mixdown path from a multitrack recorder, it allows the soloed tracks to be maintained in their normal perspective. Equalization and reverb are not defeated nor is the stereo panning control.

The solo system we are about to see will perform these functions plus provide a cue feed to the performers that is not interrupted when using the solo switches. This will allow soloing tracks even during recording and allow the existing console cue feeds to be used as additional sends. After all, wouldn't it be nice to be able to have a separate send bus for the reverb, and for the echo, *and* for the harmonizer. Since FET switches are employed for attenuation, construction is simplified (compared to, say, a bank of relays), nearly any amount of gain reduction is possible and, best of all, operation is completely silent allowing use during mixdown as a kind of "group muting." For example, let's say you're mixing a tune that has guitars and drums, etc., but has a quiet piano introduction. If you solo the piano track(s) during the beginning, you will kill any extraneous sounds on the other tracks such as guitar amp noise, chairs squeaking, punch-in thumps, etc. In fact, studios not equipped with noise reduction will find this particularly valuable as it also eliminates tape hiss from those tracks. It is usually desirable to limit attenuation to about 40 dB since the slight leakage will allow the relationship between the soloed track and the rest of the mix to be maintained.

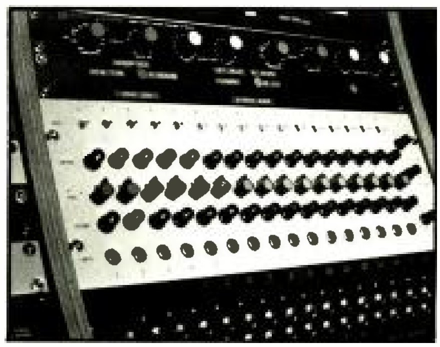


The companion cue mixer can be as simple or as elaborate as you want to make it. I incorporated all of the options on my own unit including pan pots for stereo as well as a reverb send/return system. Even if you decide to skip the separate reverb sends, you should at least consider including the return which could come from your existing reverb unit. Don't forget, it is very important to the performers that the earphone mix sound as good as possible.

You will need to make the output of the cue mixer available through the control room speakers to allow setting up a balance. Otherwise you will have to listen on earphones in order to hear what you are doing. You'll also need to determine a good place to mount the solo switches. The most obvious method would probably be to build the entire assembly onto a 19" panel and mount it in your rack. It is possible though, to actually mount the switches right on the console panel and wire directly into the console. This would allow interrupting the signal at any place in the console further expanding the possibilities. This approach would, however, also require the most bravery.

Since the output impedance of the solo circuit is relatively high, it should not be expected to drive more than about a hundred feet of cable as the capacitance of longer lengths could begin to impair high frequency performance. Likewise, if the impedance of your console's tape input is less than 5K or so, you should add some sort of line driver. The NE5534 op-amp would be

— continued overleaf . . .



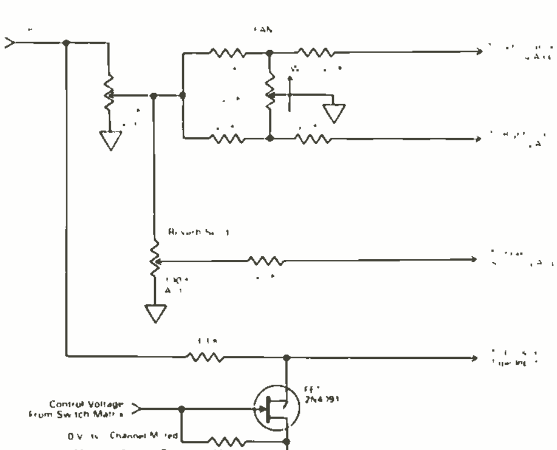


Figure 1 Schematic of One Input and FET Attenuator

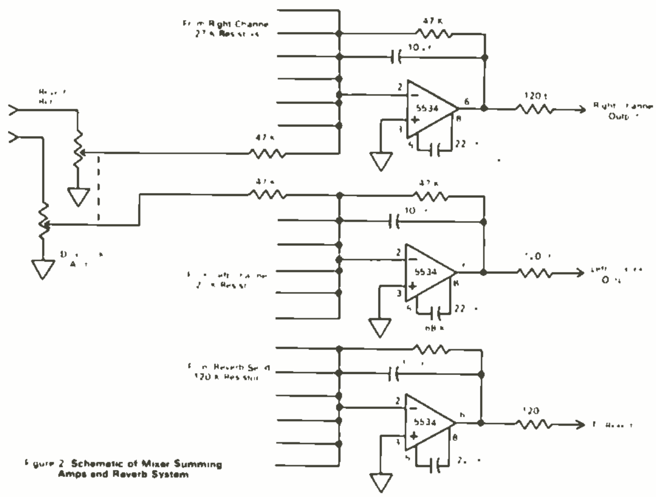
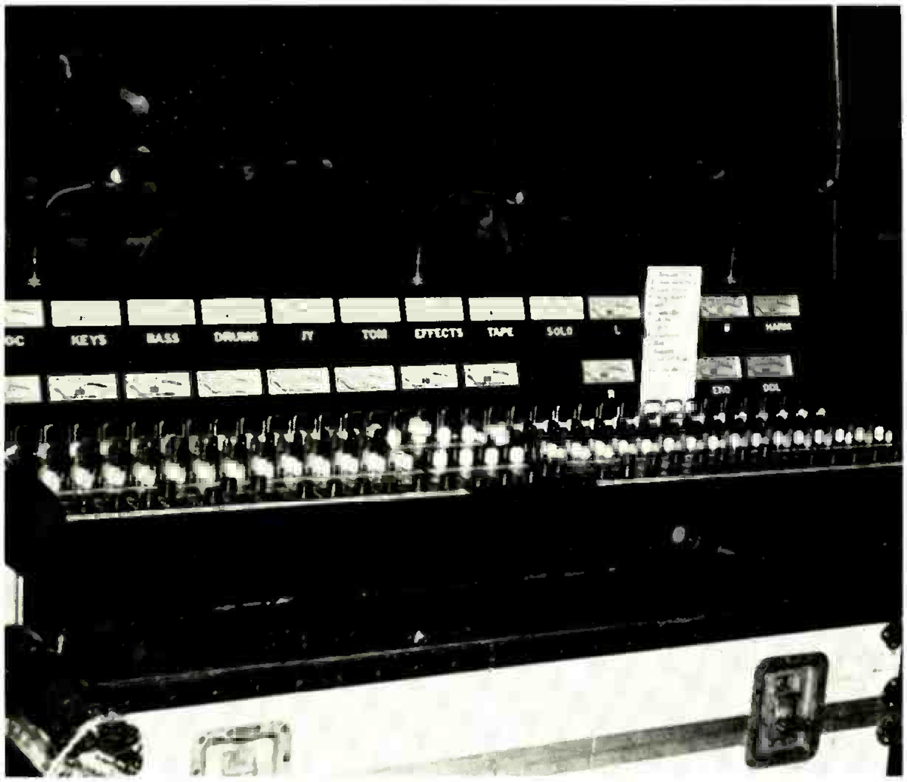
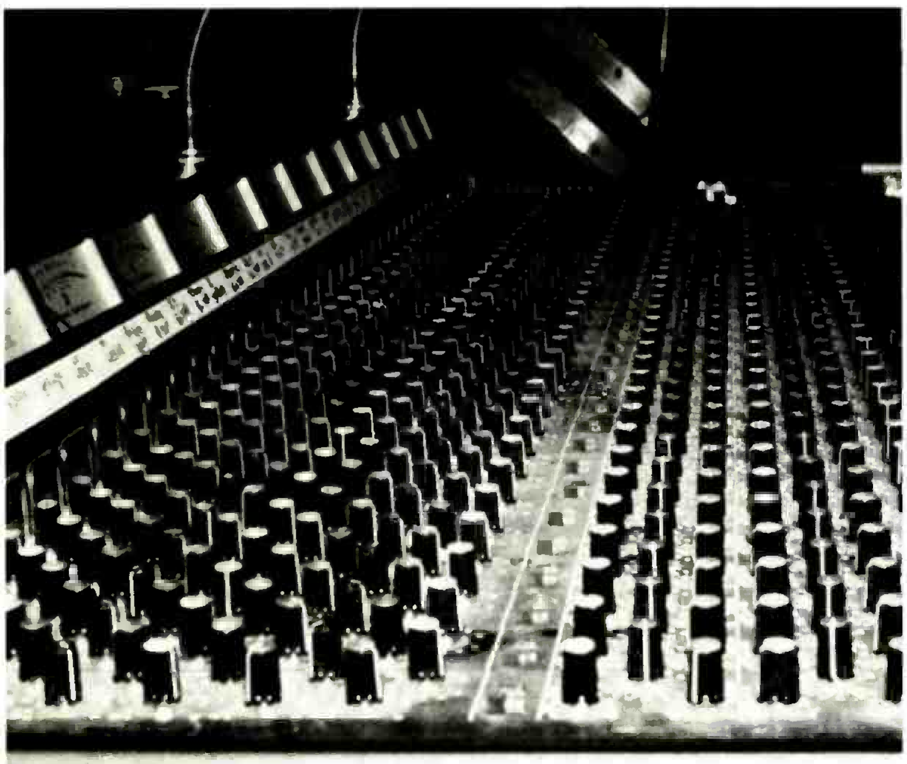


Figure 2 Schematic of Mixer Summing Amps and Reverb System



AMERICAN CONCERT SOUND DOES IT WITH INTERFACE



ROCK CONCERT HOUSE MIXERS

Model 308-32X8S-32J/NS

"House Mixers" control 32 (or more) microphones and make up to eight stereo group submixes corresponding to up to eight performer groups, each with its own submaster and observable on VU meters. The NS module then makes a stereo house mix with a constant sum house panpot and a stereo slider house master, plus an operator's mix which is pushbutton selected from one or more of the submixes or any input solo. Inputs can even be soloed when off, to check before bringing them in. The type J module is standard and the type B module with parametric equalizers is optional. Either is also available with VCA, for VCA grouping. LED bargraph VU meters on every input are another option.

These simple rugged heavy duty mixers stand up well under difficult operating conditions to give long life on the road. Modular construction and plug-in IC's permit easy servicing. Large illuminated VU meters with 30000 hour lamps are visible under all conditions. Foam lined Anvil trunks are optional.

STAGE MONITOR MIXERS

Model 104-32X4A-32L/NS
312-32X12A32L/NS

Stage Monitor mixers make a number of mixes for feed back to a number of performer groups on stage, each of which wants a different mix. This requires a direct matrix and cannot be done with a standard pushbutton mixer. The 104L module has input pad, gain set switch, three equalizers, four position rolloff, LED overload danger indicator and solo, similar to the 104J, but the slider attenuator is replaced by eight color coded send pots, each with an eq in/out switch. The mixer thus makes a 32 X 8 pot matrix from inputs to outputs. Pot masters are standard, sliders optional. The NS operator's monitor permits the operator to listen to any mix or to any input solo.

These rugged, reliable Stage Monitors have become the standard of the industry and are used by many professional sound companies.

The 312L Stage Monitor is similar but adds three parametric equalizers, 12 send pots rather than 8, and a four inch slider attenuator to the input module, and is in a 308 frame. Slider Masters are an optional extra. The 312L system makes 12 output mixes.

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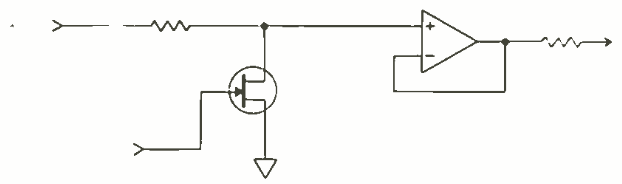

Figure 3 FET Attenuator With Optional Line Driver

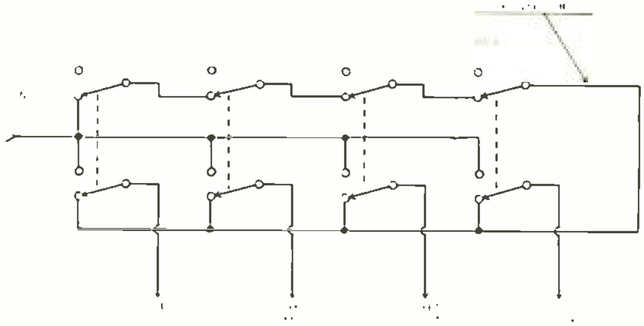

Figure 4 Solo Switching Matrix

ideal for this application though when dealing with a lot of tracks, it can become expensive. When headroom is not a problem, other, less expensive ICs may be substituted. These include the LF356, the RC4136 quad, and the TL074 and TL084 quads with FET inputs. The quads have four independent op-amps in one package sharing only the power supply connections. If you use the 5534 as a line driver, be sure to install a 22 pf capacitor between pins 5 and 8 for stability. The cue mixer is capable of driving nearly any cable length and impedances down to 600 ohms without any trouble. But to get back to the solo circuit, the FET switches are employed in a "shunt" configuration, activated only when required for muting. This means that no noise or distortion is ever added when a given track is being used. Even when muted, distortion will be well under 1% of the *muted* level. In other words, at least 80 dB down!

Since the solo switches carry only control voltages and not the actual audio, shielded wiring is not necessary. This also means that the switches can be mounted anywhere without regard to distance from the rest of the circuit or being in proximity to AC wiring or hum fields. Toggle switches or latching pushbutton types are preferred over the momentary kind. Otherwise you're likely to find yourself standing there trying to hold down a button with one hand while patching in a limiter or tweaking an equalizer with the other.

The cue mixer circuit is relatively straightforward. Referring to Figure 1, each input is fed from one of the multitrack tape deck's outputs, though before the solo attenuator. An audio taper pot is used to control the level, then the signal is split into two with another pot establishing the balance. The reverb send is also taken at this point with another control being used to determine the send level. Figure 2 shows how the three groups — left, right, and reverb — are then combined, each at its own "summing amp." Many inputs can be mixed together this way without any interaction of the controls. (For a complete overview of state-of-the-art summing methods, refer to the excellent article on console design by John Roberts in the April, 1980 issue of *R-e/p.*) Finally, the reverb return is mixed back in along with the rest of the inputs in stereo with a control provided for return level.

The FET switching is simple enough to understand — when the control voltage at the gate terminal is very small, the transistors impedance between source and drain becomes quite low shunting most of the audio signal to ground. When the voltage rises above approximately 5 volts, the FET goes through a transition and changes to a much higher resistance. It is important to note that the FET selected for this project requires a *negative* bias voltage, though the principle is still the same. To help illustrate the concept, an FET attenuator is shown by itself in Figure 3. Also shown is the optional line driver circuit that is required when connecting to low impedance inputs.

The actual mechanical switching circuit

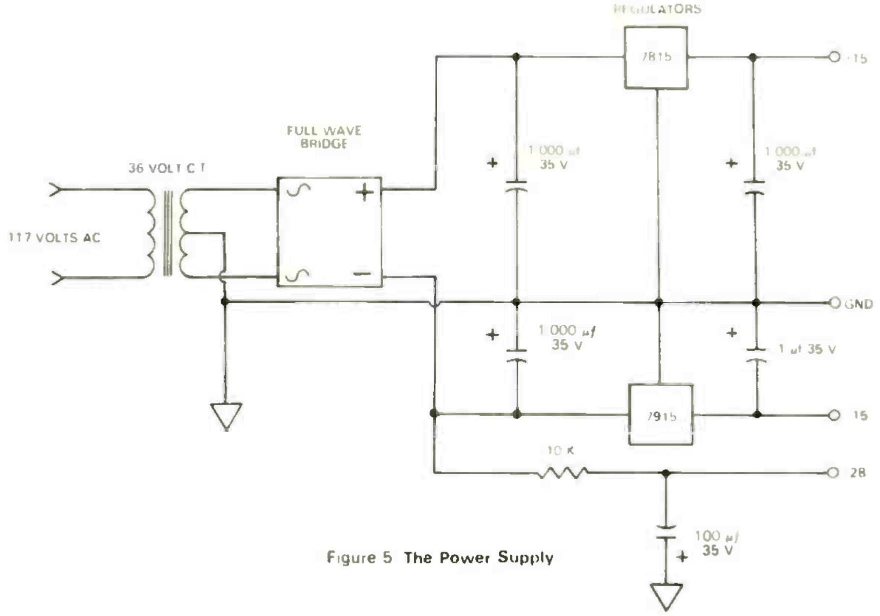

Figure 5 The Power Supply

shown in Figure 4 is also fairly simple, though perhaps a brief explanation is in order. Each switch is a double-pole type since two different things have to happen when it is activated. One-half of each switch is connected in a chain configuration requiring all of the switches to be in the "normal" position in order for the control voltage to get through to the FET gates. Activating any switch interrupts the voltage to all of the FETs. At the same time, the control voltage for the FET associated with that switch is routed around this "interruptable" bus coming instead directly from the -28 volt supply. Using this arrangement it is possible to solo as many channels as desired at one time. While only four switches are shown for the sake of simplicity, this system will obviously work with any number required. The cue mixer also can be wired to accommodate as many inputs as desired and, in fact, can be used as the basis for a high quality mixer for any line level application. □□□

PARTS LIST

Power Supply

1 - Power Transformer, 36-volt center-tapped, 65 mA.
1 - 10K ¼-watt, 5% resistor.
2 - 1 uf 35-volt tantalum capacitors.
1 - 100 uf 35-volt electrolytic capacitor.
2 - 1,000 uf 35-volt electrolytic capacitors.
1 - 7815 positive voltage regulator.
1 - 79M15 negative voltage regulator.

Mixer Electronics

3 - 120-ohm ¼-watt, 5% resistors.
4 - 47K ¼-watt, 5% resistors.
1 - 68K ¼-watt, 5% resistor.
1 - Dual 10K audio taper potentiometer.
3 - 10 pf capacitors, mica or disc ceramic
3 - 22 pf capacitors, mica or disc ceramic
3 - NE5534N op amps, Signetics.

Per Channel

1 - DPDT switch.
1 - 3.3K ¼-watt, 5% resistor.
4 - 27K ¼-watt, 5% resistor.
1 - 120K ¼-watt, 5% resistor.
1 - 330K ¼-watt, 5% resistor.
1 - 10K audio taper potentiometer.
1 - 100K audio taper potentiometer.
1 - 20K linear taper potentiometer.
1 - 2N4091 N-channel FET.

***The following components** are available postpaid from* Phoenix Systems, 375 Springhill Road, Monroe, Connecticut 06468:

36-volt transformer #P-1220-T	$5.00
NE5534N op-amp	$3.50
TL074 quad op-amp	$2.50
7815 positive regulator	$1.50
79M15 negative regulator	$2.50
2N4091 FET	$1.50

A $1.00 handling charge must be added to orders of less than $10.00.

Automatic Microphone Mixing

by Chris Foreman

Versatility vs. Complexity, the Dilemma

Electronics technology continues to make major changes in the audio marketplace. Compared to just a decade ago, a potential buyer will find not only improved performance, but a bewildering array of new features, even completely new products.

For the audio professional (with lots of money to spend) it must be like being a kid in the world's largest toy store! Everything is "computer-equipped" and "digitally-controlled." Performance ratings extend to the nth decimal point.

But, as we all know, every silver lining has a cloud. As the kid in the toy store (audio pro) soon learns: 1) It's hard to make choices when everything looks so good. 2) When you finally get the stuff home, you've got to spend a day reading the instruction manuals before you can even turn it on!

Worse yet, consider the non-technical user of audio equipment. A music-lover and would-be hi-fi buyer, for example, needs a course in audiophile jargon before setting foot in most stereo shops. And in commercial sound, from courtrooms to airports, users have a hard time benefiting from new technology. These people are often professionals, but in a field completely unrelated to audio. They have neither the time nor the desire to learn the functions of dozens of controls and switches.

The point is, the increased versatility associated with new technology often leads to increased complexity. In many cases even professional users find this increased complexity burdensome. For non-technical users, operational complexity can actually be an impediment to full utilization of the equipment.

What can be done? One answer is to transfer the burden of complexity away from the user, and back to the manufacturer. Automated features are the best example of this transfer. The hi-fi buyer can now purchase a cassette deck that automatically adjusts bias and EQ for the user's choice of tape. Fewer knobs to twiddle, but increased performance from the hi-fi system. For commercial sound users, Altec Lansing introduced the Dugan system "Automatic Microphone Mixer" in 1976. Its front panel looks like a conventional commercial sound mixer; nothing to frighten a potential user. Yet, the Dugan/Altec automatic mixer actually "mixes" individual microphone levels much like a trained human operator would mix them. It raises the level of in-use microphones and lowers the level of others, thus reducing ambient noise pickup. In addition, it keeps overall system gain constant, dramatically lowering the possibility of acoustic feedback.

Thus, the automatic mixer makes life a little easier for the end user. What about the system designer? The installer? Well, the news isn't quite as good, but it's not bad either. As the high-technology products go, the automatic mixer is one of the easiest to understand and apply. This article explains how an automatic mixer works, and how to design sound systems around it. I've included examples for commercial sound and entertainment-oriented systems.

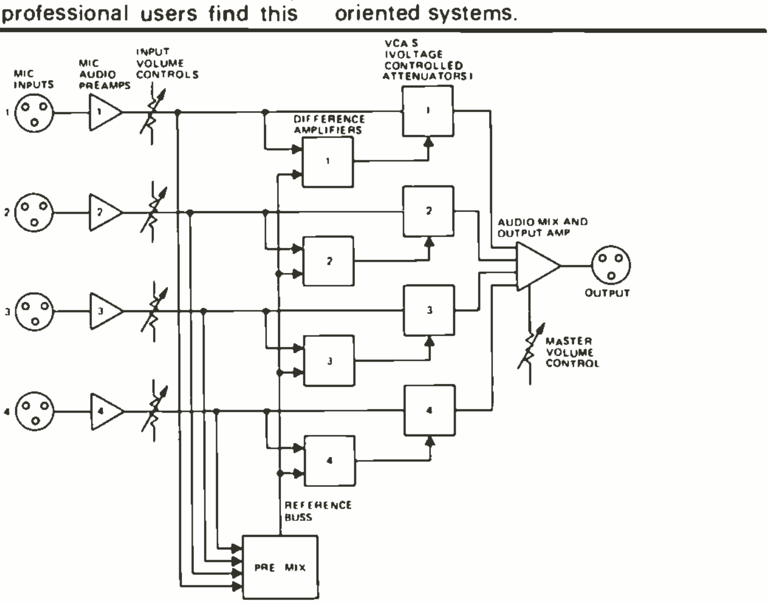

Figure 1. Simplified Block Diagram of a Four-Input Automatic Mixer

The Altec/Dugan System

This article is based on the Altec/Dugan system.[1] Several other manufacturers now offer commercial mixers with automated features. The applications described in this article should apply, at least in part, to those other mixers. The "How It Works" section, however, applies exclusively to the Altec/Dugan system.

How It Works

Refer to Figure 1, a hypothetical four-input automatic mixer. With the exception of the analog computer circuits, indicated by squares, this could be the block diagram of a conventional four-input mixer. These analog computer circuits perform the automatic mixing functions according to one simple rule: **Each individual input channel is attenuated by an amount, in dB, equal to the difference, in dB, between that channel's level and the sum of all channel levels.**

This rule can be stated mathematically as follows:

$$L_n' = L_n - [\mathrm{Sum}(L_n) - L_n]$$

Where:

L_n is the level in channel n before attenuation.

L_n' is the level in channel n after attenuation.

$\mathrm{Sum}(L_n)$ is the sum of the levels in all channels (the sum is taken before the individual channels are attentuated).

All values are in dB notation.

Notes

(1) In the block diagram, L_n is the level in any channel immediately following that channel's volume control. Note that the volume control allows the human operator to adjust the level in that channel **before** any automatic mixing takes place.

(2) The square labeled "Pre Mix" sums the levels from all channels (after their volume controls) and produces the term $\mathrm{Sum}(L_n)$ in the equation.

(3) The output of the Pre Mix circuit feeds the square labeled "Difference Amplifier" in each input channel. The Difference Amplifier circuit in each channel performs a linear subtraction function producing the term $[\mathrm{Sum}(L_n) - L_n]$ in the equation.

(4) The "VCA" (voltage-controlled attenuator) circuit now performs the final automatic operation by subtracting the bracketed term from the original signal. The output of the VCA is the final input channel level, L_n'.

(5) The L_n' levels from all channels are finally mixed and pass through the Master Volume Control to the output of the mixer.

continued overleaf . . .

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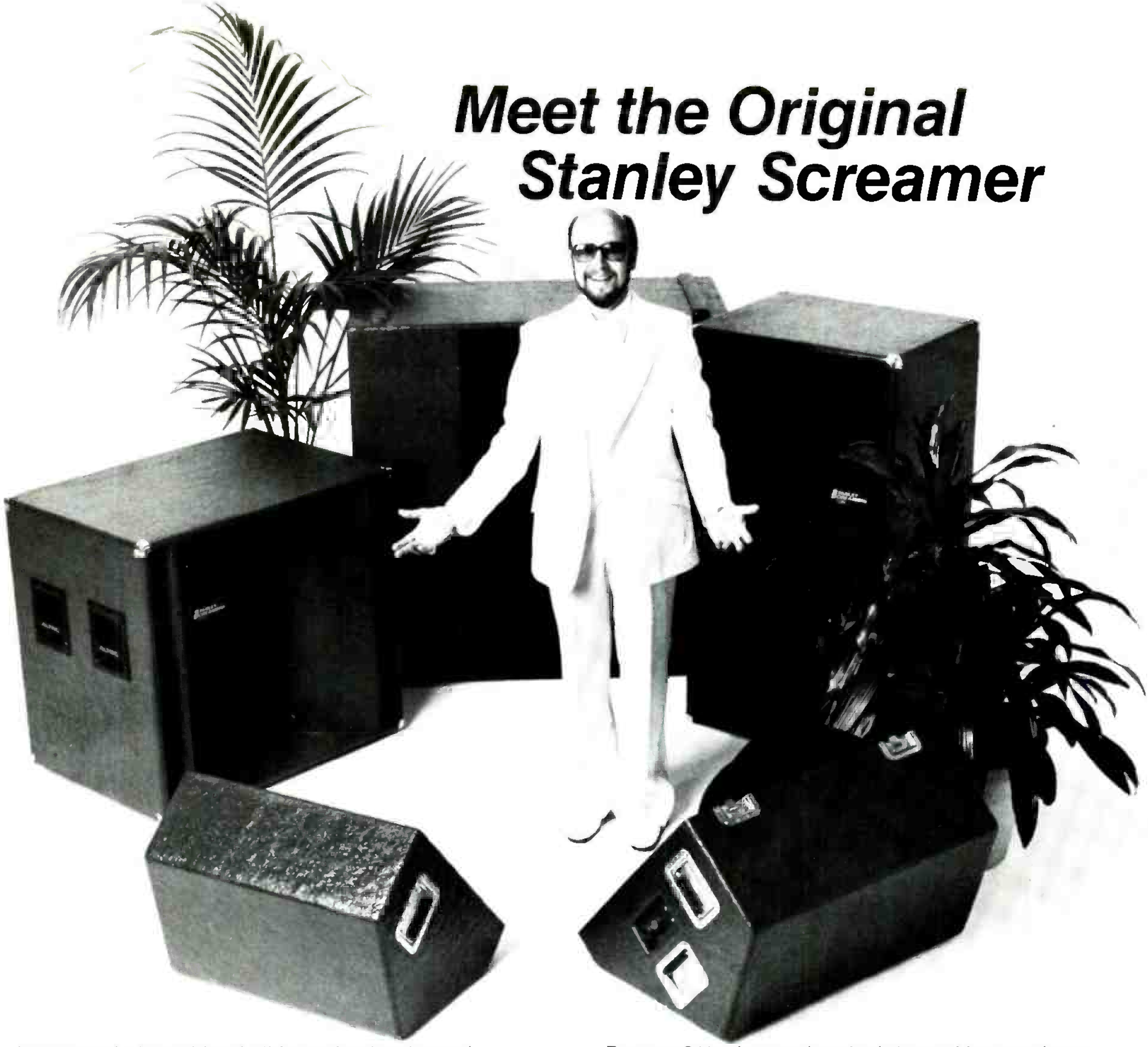

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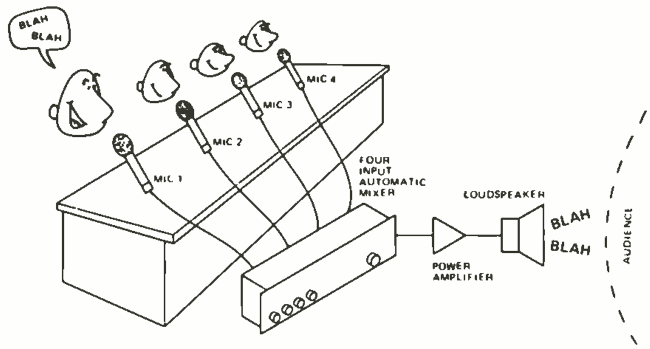


Figure 2A. Simplified Four-Input Automatic Mixer in a Sound System With One Microphone In Use

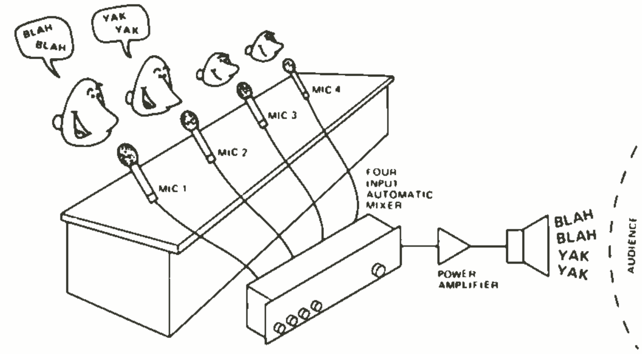


Figure 2B. Simplified Four-Input Automatic Mixer in a Sound System With Two Microphones In Use

An Example

Now install this simplified automatic mixer in a hypothetical sound system, as shown in Figure 2A. Since only Mike 1 is being used, none of the other microphones are contributing to $Sum(L_n)$, the Reference Bus Level. Thus, $Sum(L_n)$ is equal to L_1, the Channel 1 Preamp level. For Channel 1, this sets $L_1' = L_1$, and Channel 1 is not attenuated. For Channel 2, $Sum(L_n)$ is much larger than L_2. Thus, for Channel 2, the equation becomes $L_2' << L_2$, that is, Channel 2 is attenuated greatly. In a similar manner, Channels 3 and 4 are also attenuated greatly and are effectively "off."

Next assume that Channel 1 and Channel 2 microphones are both in use at equal input levels as shown in Figure 2B. Now $Sum(L_n)$ is equal to the sum of the two levels. In dB notation this means that $Sum(L_n)$ is 3 dB higher than either L_1 or L_2. Thus, for Channel 1, the equation becomes $L_1' = L_1 - 3$ dB.

Channel 2 is also attenuated by 3 dB. Channels 3 and 4, however, are attenuated greatly as in the first example.

After attenuation, the individual channels are mixed together and routed to the main mixer ouput. These two microphone levels, each attenuated by 3 dB, mix back together to produce a signal level that is exactly equal to the level of either microphone before attenuation. Thus, the final level is the same as if only a single microphone had been in operation. This action is called "gain-sharing."

Operator Interface

Even on this simplified automatic mixer, there are manual volume controls for each input channel. These controls allow the operator to adjust the level of the input channels **before any automatic mixing takes place.** For example, the operator could raise the level of one channel and lower the level of another channel to compensate for the difference between a weak-voiced talker and a strong-voiced talker.

Coherent and Non-Coherent Signals

In the previous example, different talkers used two different microphones and the signals entering these two microphones were totally unrelated to each other. Signals like these, which bear no relation to each other, are called "non-coherent" signals.

A single talker, positioned an equal distance from the two microphones, would have produced an equal signal in both microphones. Signals like this are called "coherent" signals. Coherent signals don't have to be equal in level, but do have to be very similar to each other. Another example of coherent signals reaching two or more microphones results when a door is slammed or a book dropped at an approximately equal distance from two or more microphones.

The significance of coherent and non-coherent signals is this: When two **non-coherent** signals of equal level are mixed together, the resultant signal is **3 dB** higher than either of the two original signals. When two **coherent** signals of equal level are mixed together, the resultant signal is **6 dB** higher than either of the two original signals.

The Altec/Dugan system automatically compensates for the difference between coherent and non-coherent signals, a natural result of the system's operational equation. This helps the system avoid potential mixing errors in the case of the slammed door or dropped book. Without this ability, the mixer could conceivably allow the sound system to go into acoustic feedback when high-level coherent signals were encountered.

APPLICATIONS FOR THE AUTOMATIC MIXER

An automatic microphone mixer can improve most commercial sound systems and many entertainment-oriented sound systems. It helps avoid acoustic feedback in multi-microphone systems. It automatically selects "in-use" microphones and attenuates others. In commercial

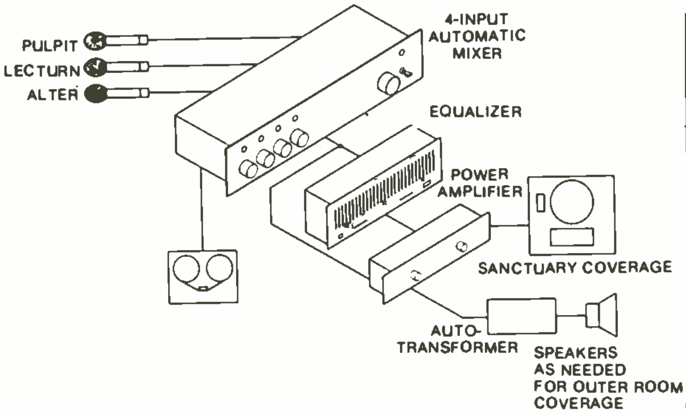


FIGURE 3. A SMALL CHURCH SOUND SYSTME USING A FOUR-INPUT AUTOMATIC MIXER

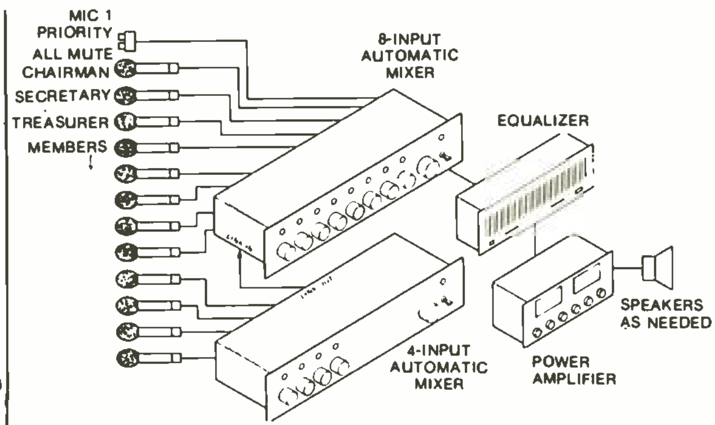


FIGURE 4. A 12 MICROPHONE MEETING ROOM SYSTEM USING LINKED 4- and 8-INPUT AUTOMATIC MIXERS

sound systems, this can result in total "hands-off" mixing where the only control the user needs to operate is the on-off switch. In entertainment-oriented systems, the automatic mixer is a **mixing aid** which frees the operator to attend to the "aesthetics" of sound system operation.

A Church Sound System
(Refer to Figure 3)

Most church sound systems have several microphones. Often, however, only one microphone is in use at any given time. To complicate this situation, many churches have several services each Sabbath, and various classes and meetings during the week. Often, a different person is responsible for sound system operation for each event.

For these reasons, an automatic microphone mixer can provide a valuable service in church sound systems. Besides helping to avoid feedback and reducing ambient noise pickup, the automatic mixer helps maintain consistent system operation even when different people operate the system at different services.

A Meeting Room System
(Refer to Figure 4)

Meeting room sound systems include company board room and conference room systems, courtroom systems, and systems for legislative bodies or other large groups. In these systems, several automatic mixers may be "linked" to build an automatically mixed system with many microphones (up to 48 microphones with the Altec/Dugan system). In the system shown, the first microphone position includes a Mike 1 Priority and an All-Mute switch.[2] The mike 1 priority feature mutes all microphones except mike 1. This feature can be a valuable aid in keeping order during a meeting. The all-mute feature mutes all microphones, including mike 1.

The mike 1 priority and all-mute features are important, but the primary advantage of an automatic mixer in a meeting room system is in helping to avoid feedback. Every time the number of in-use microphones in a system doubles, the sound system moves 3 dB closer to feedback. Since the automatic mixer automatically attenuates un-used microphones and shares the gain between in-use microphones, the feedback potential is greatly reduced.

A Courtroom System
(Refer to Figure 5)

The automatic mixer can eliminate many problems associated with traditional courtroom sound systems. The mike 1 priority feature allows the judge to mute all microphones but their own giving them an extra measure of control over courtroom proceedings. The all-mute feature allows the judge to mute all microphones in order to hold private conversations at the bench.

Many courtroom systems include a multi-channel "logging" tape recorder. Some automatic mixers, including the Altec/Dugan mixers, provide a line output from **each microphone input** so that each microphone can be recorded on a separate channel of the logging recorder. The line outputs are not affected by the front panel controls nor are they affected by actions of the automatic mixing circuitry.[2] Thus, any inputs to the microphones are recorded, just as they occurred. Even if the judge activates the mike 1 priority feature, all microphones will be recorded. Should the judge activate the all-mute feature, however, no microphones will be recorded, preserving the privacy of conversations even on the logging recorder.

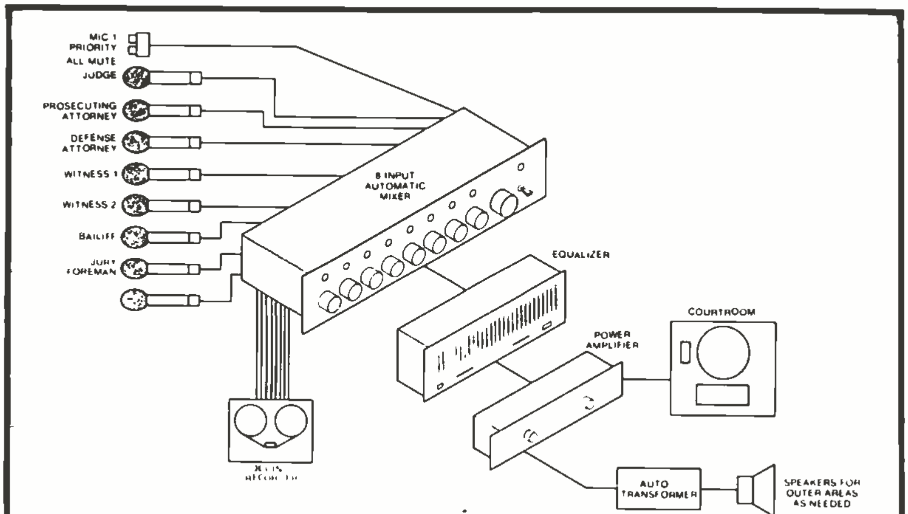


Figure 5 A COURTROOM SYSTEM USING AN EIGHT—INPUT AUTOMATIC MIXER

Entertainment Systems
(Refer to Figure 6)

The uses of an automatic mixer are not limited to commercial sound systems. An automatic mixer can be invaluable in a live drama presentation where multiple "footlight" and behind-scenery microphones are used. The automatic mixer selects the microphone closest to the performer(s) talking (or singing) and reduces the level of all other microphones. As these microphones are usually hidden, this operation is extremely difficult for an operator unaided by an automatic mixer.

The automatic mixer may also be useful for recording or reinforcing trap (drum) sets. There are normally several microphones on a trap set, and, especially during recording sessions, it would be ideal to have only one microphone pick up the action at any one time, thus minimizing phase cancellations between mutltiple open microphones. Unfortunately, a single microphone cannot be properly placed to pick up the entire trap set, and multitple microphones must be used. That's where the automatic mixer comes into play. By automatically attenuating those microphones which are farther away from the (moving) acoustic source, the automatic mixer reduces phase cancellations. This should result in a "tighter," more accurate drum sound. During a live concert, the automatic mixer would also reduce the possibility of feedback from the multiple open microphones used on the trap set.

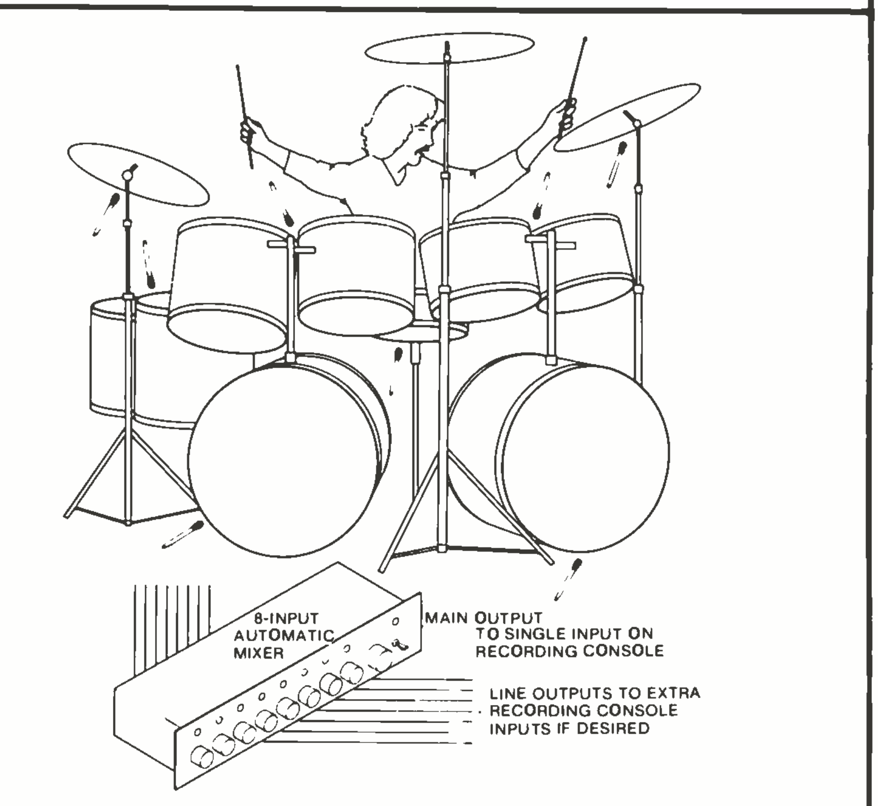


Figure 6: RECORDING A TRAP SET WITH AN EIGHT-INPUT AUTOMATIC MIXER

Author's Note

While I have personal experience with automatic mixers in many of the systems described in this article, I have never tried an automatic mixer on a drum kit! The theory is sound, the experience is lacking. Since this application could be valuable to many readers of R-e/p, (and I'd certainly like to know how well it works), I urge anyone with such experience to write R-e/p and report.

When It Doesn't Work

An automatic mixer is not very useful in mixing group vocals. If one member of the group sings louder than the others, the automatic mixer reduces the level of the other microphones. Thus, for this application, the automatic mixer mixes "backwards."

In some cases, however, this could be a desirable action. In a large choir, for example, the automatic mixer would keep the various microphone levels relatively constant until a soloist approached their microphone. At this point, the automatic mixer would attenuate the levels of the other microphones, allowing the soloist to predominate in the mix. As the soloist finished, the automatic mixer would allow the other microphones to return to their previous levels.

For this same reason, the automatic mixer could be useful in any vocal or instrumental group where the musicians are relatively disciplined (no one attempts to be a soloist except when the music calls for a soloist!), yet it is desirable to allow a soloist to predominate during certain passages.

Broadcast System

Radio and TV talk shows, interview segments, and newscasts can benefit from the automatic mixer. How often have you seen a newscaster come on camera without sound! We often blame this on the sound mixer (human operator), yet it is extremely difficult to know who is going to talk during the free-for-all conversations that sometimes develop. An automatic mixer would raise the level of the appropriate microphones and reduce the level of other microphones. This would reduce ambient noise pick up from microphones not in-use, yet allow every important word to be broadcast.

Using The Logic Outputs

The "Logic Output" feature available on some automatic mixers, including the Altec/Dugan system, greatly extends their usefulness in many systems. The logic outputs may be connected to switching systems for such purposes as switching loudspeakers on and off, controlling tape recorders or other devices, or for implementing complex priority systems.

If they are TTL compatible, the logic outputs may be connected directly to a properly designed TTL or CMOS switching circuit (these external circuits must have their own DC power supply). For a detailed explanation of the design and uses of TTL switching circuits, see the "TTL Cookbook," by Don Lancaster. Don Lancaster has also written the "CMOS Cookbook." Both are published by Howard W. Sams Publishing Company. A word of caution: consider grounding carefully when connecting the logic outputs to an external TTL or CMOS circuit to avoid potential ground loops.

TTL compatible logic outputs may be used to drive low-power relays, requiring less than 5 milliamps of current to activate their coils. For relays requiring higher coil currents, use an external power supply of appropriate voltage and a circuit such as the one shown in Figure 7.

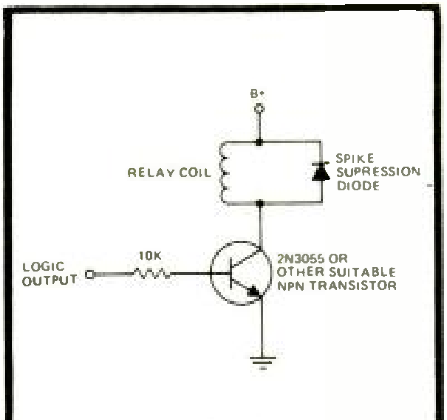


Figure 7

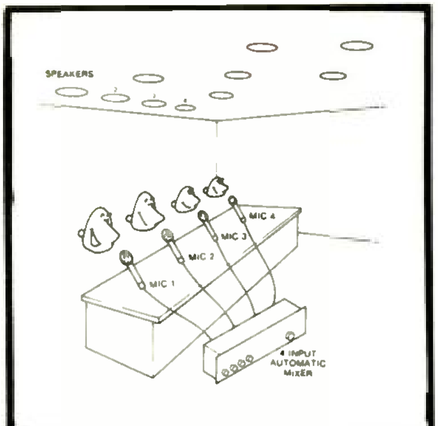


Figure 8 ADDING ADDITIONAL FEEDBACK STABILITY MARGIN BY USING THE AUTOMATIC MIXER'S LOGIC OUTPUTS

Extra Feedback Stability Margin In Conference System
(Refer to Figure 8)

In conference systems using distributed loudspeaker systems, individual microphones are often located directly under a ceiling-mounted loudspeaker. Obviously, the feedback potential is high. Using the logic outputs to disconnect the appropriate loudspeaker decreases this feedback potential. In Figure 8, logic output #1 activates a relay which disconnects loudspeaker #1 (directly over microphone #1) whenever someone is talking into microphone #1. When microphone #1

— *continued overleaf* . . .

Automatic Microphone Mixing

by Chris Foreman

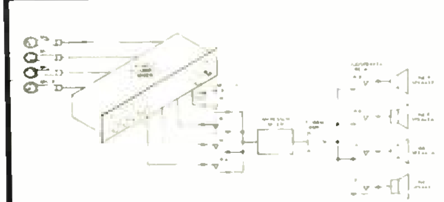


FIGURE 9A AN AUTOMATIC ZONE PAGING SYSTEM USING AUTOMATIC MIXER LOGIC OUTPUTS

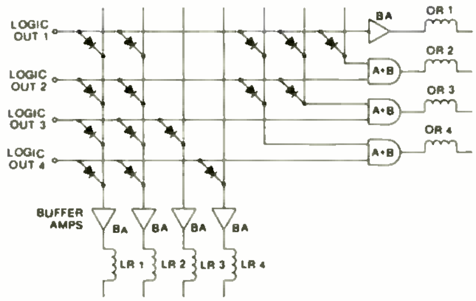


Figure 9B: LOGIC FLOW DIAGRAM FOR THE AUTOMATIC ZONE PAGING SYSTEM

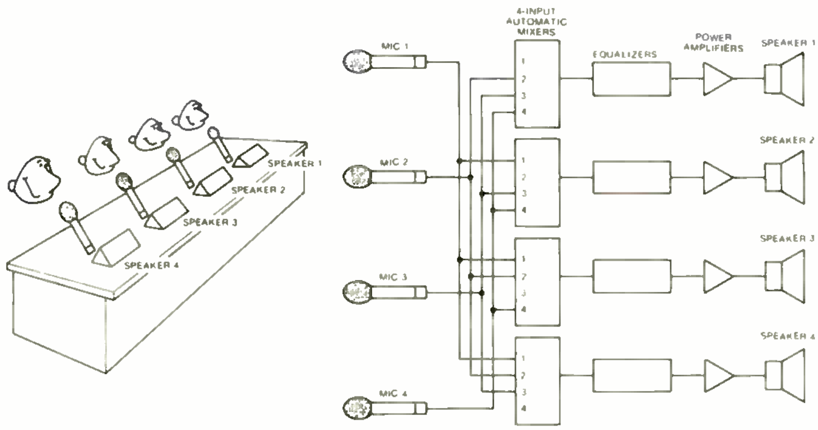


Figure 10A: "MIX-MINUS-SELF" MONITORING NORMAL IMPLEMENTATION

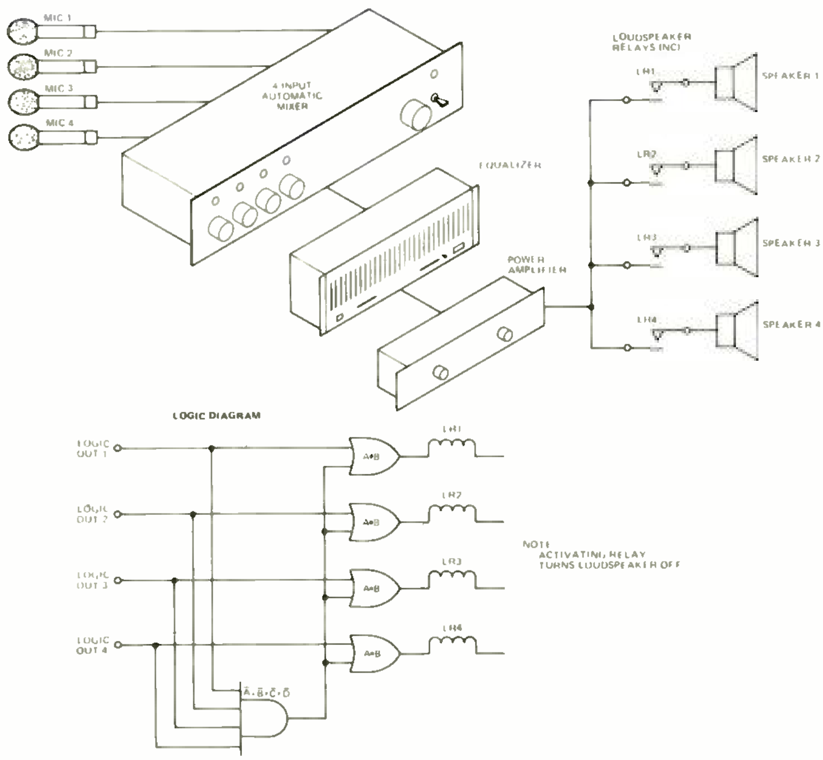


Figure 10B: "MIX-MINUS-SELF" USING AUTOMATIC MIXER'S LOGIC OUTPUTS

is not in use, it is automatically attenuated by the automatic mixer. Thus, feedback potential is reduced yet the loudspeaker retains its usefulness as a monitor to anyone seated nearby.

All four logic outputs are used to switch off the appropriate loudspeakers. In addition, all four loudspeakers are switched off whenever all four microphones are quiet. This reduces amplification of background noise.

Automatic Zone Paging And Microphone Priority Switching

(Refer to Figures 9A and 9B)

This example illustrates the use of the logic outputs to implement both automatic zone paging and an automatic microphone priority system. In many systems, only one of these functions would be needed.

In a conventional zone paging system, a switch on a push-to-talk microphone activates both the microphone itself and, via an extra wire in the microphone cable, a set of relays to select the appropriate loudspeaker zone(s). In the example shown, the microphone switch activates the microphone only. The loudspeaker relays are operated via the logic output associated with that microphone. This eliminates the need for the extra wire and DC voltage in the microphone cable, a common source of noise.

The purpose of a microphone priority system is to allow emergency paging to automatically mute any other page that may be taking place. In Figure 9B, the logic outputs are used to implement both the automatic microphone priority system and the automatic zone paging. The diode circuit shown is not necessarily a workable circuit and is given for illustration only.

"Mix-Minus-Self" Monitoring[3]

(Refer to Figures 10A and 10B)

The purpose of mix-minus-self monitoring is to allow persons seated at a long conference table to hear all talkers except themselves. Figure 10A shows the conventional method of implementing a mix-minus-self system using multiple automatic mixers, equalizers and powerful amplifiers. This system can be extremely successful but requires a large quantity of equipment.

Figure 10B shows a mix-minus-self system using the logic outputs of an automatic mixer. This implementation requires fewer mixers, equalizers, and power amplifiers. Instead of mixing each talker's loudspeaker separately, this system simply disconnects each talker's loudspeaker when they are talking (similar to the system of Figure 9).

The disadvantage of this system is that during a heated discussion, when two talkers are talking simultaneously, both talker's loudspeakers may be disconnected part of the time, preventing both talkers from hearing each other. (This might be an advantage.)

Planning A System For Automatic Mixing

In general, there are no changes in design philosophy for systems using automatic mixers. There are a few "tricks," however:

(1) Let NOM = 1 (NOM is "Number of Open Microphones as used in accepted sound system formulas). This applies regardless of the number of microphones used since the automatic mixer will force multiple open microphones to share the overall gain of the system.

(2) Chose D_s (talker to microphone distance), D_1 (Microphone to loudspeaker distance), D_2 (loudspeaker to listener distance) and D_0 (talker to listener distance) for the worst-case microphone. If the sound system will work with this worst-case microphone, the automatic mixer will allow the system to work with multiple microphones with no increase in feedback potential.

(3) Experience with automatically mixed systems indicates that D_s is usually longer in a typical automatically mixed system than it is in a conventional system. This means the output from the microphone is lower, and electrical signal-to-noise ratios are degraded. A high-quality, high-output microphone will reduce system noise and provide higher electrical output to help keep the system signal-to-noise at an optimum.

(4) Avoid the use of compression in an automatically mixed system. Both compression and automatic mixing are forms of automatic gain control. Unfortunately, a compressor tends to "un-do" the automatic mixing actions and can cause an unnatural sounding system. A limiter, on the other hand, set up to provide protection from extremely high input levels, is a valuable addition to any sound system, including an automatically mixed system. Set the "threshold" at a high enough level to avoid limiting actions during normal system operation. Set the "compression ratio" at a high enough setting to adequately protect the system.

And

The automatic mixer is an exciting new tool. Yet it's not really all that new. The Altec/Dugan system can boast of almost five years of field experience and literally thousands of installations. The concept works.

So, design, install and use your automatically mixed system with confidence. A little extra homework on the part of the system designer and installer can provide significant benefits for the end user. □ □ □

Notes and Credits:

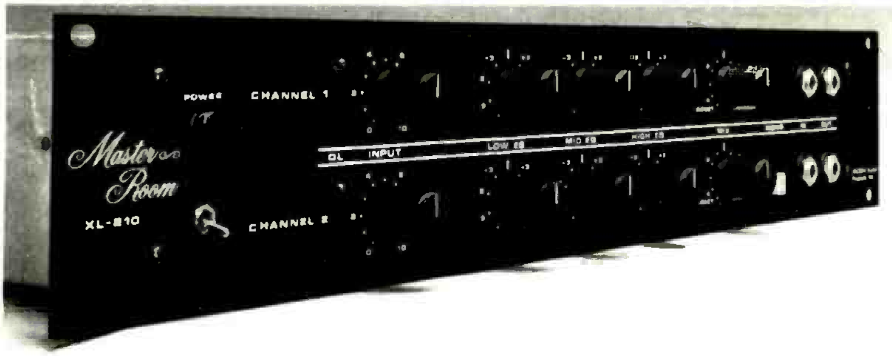
1 - U.S. Patent #3,992,584.

2 - Applies to the Altec/Dugan system but not necessarily to other automatic mixer systems.

3 - The idea for this system was presented to me by Ken Fause, of Fause & Associates.

4 - Most of the theory in this article is adapted from an AES paper by Dan Dugan, inventor of the Altec/Dugan system.

New Products



MASTER-ROOM XL-210 REVERBERATION SYSTEM

The Master-Room XL-210 incorporates recent technological breakthrough that no provides the highest quality reverberation at a affordable price. For many years, it has been th desire of professional users to have a truly hig quality reverberation system for unde $1,000.00. This long-awated desire is now reality with the XL-210.

This new technology (patent applied for) wa first utilized in the Master-Room XL-305. Base on the same technology, the XL-210 is said t provide an extremely smooth and natura sound, even on the most demanding percussiv material. According to the manufacturer, th XL-210 produces none of the unwanted sound such as boing, twang, and flutter that are common to most spring-type systems. This outstanding performance is achieved without utilizing internal limiting or any other signal manipulation methods intended to compensate for such previous system deficiences.

The XL-210 is a self-contained 3½-inch rack mount unit that features two completely independent stereo channels that are easily switchable to monaural operation. Input and output connections are via ¼-inch phone jacks located on both the front and rear panels. This unique feature allows convenient break-in patching at the front panel without disturbing the permanent rear panel connections. Active balanced inputs allow the unit to be easily fed by either balanced or unbalanced lines, and the unbalanced outputs will readily drive a 600 ohm load. The XL-210 can be used with the echo or effects send/return function of most consoles or can be placed in the main signal path, blending the desired amount of direct and reverberated signal with the front panel MIX control.

Both channels of the XL-210 feature an equalization section that provides a great deal of flexibility and creative freedom. This EQ allows the user to effectively simulate the reverberant sounds of a live chamber, plate, or concert hall. The Master-Room XL-210 incoporates special chamber isolation techniques that allow the system to be located near loudspeakers operating at high levels without acoustic feedback. The system is ruggedly built to withstand the rigors of road use, and will operate on either 120 or 240 volts. Suggested user price is $950.00.

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

for additional information circle no. 61

dbx® 900 SERIES MODULAR SIGNAL PROCESSING SYSTEM INTRODUCED

The dbx 900 Series includes a noise gate, a de-esser, and compressor, introduced by dbx, Inc. According to Lawrence Jaffe, dbx Director of Marketing and Professional Sales, "the 900 Series features convenience with flexibility."

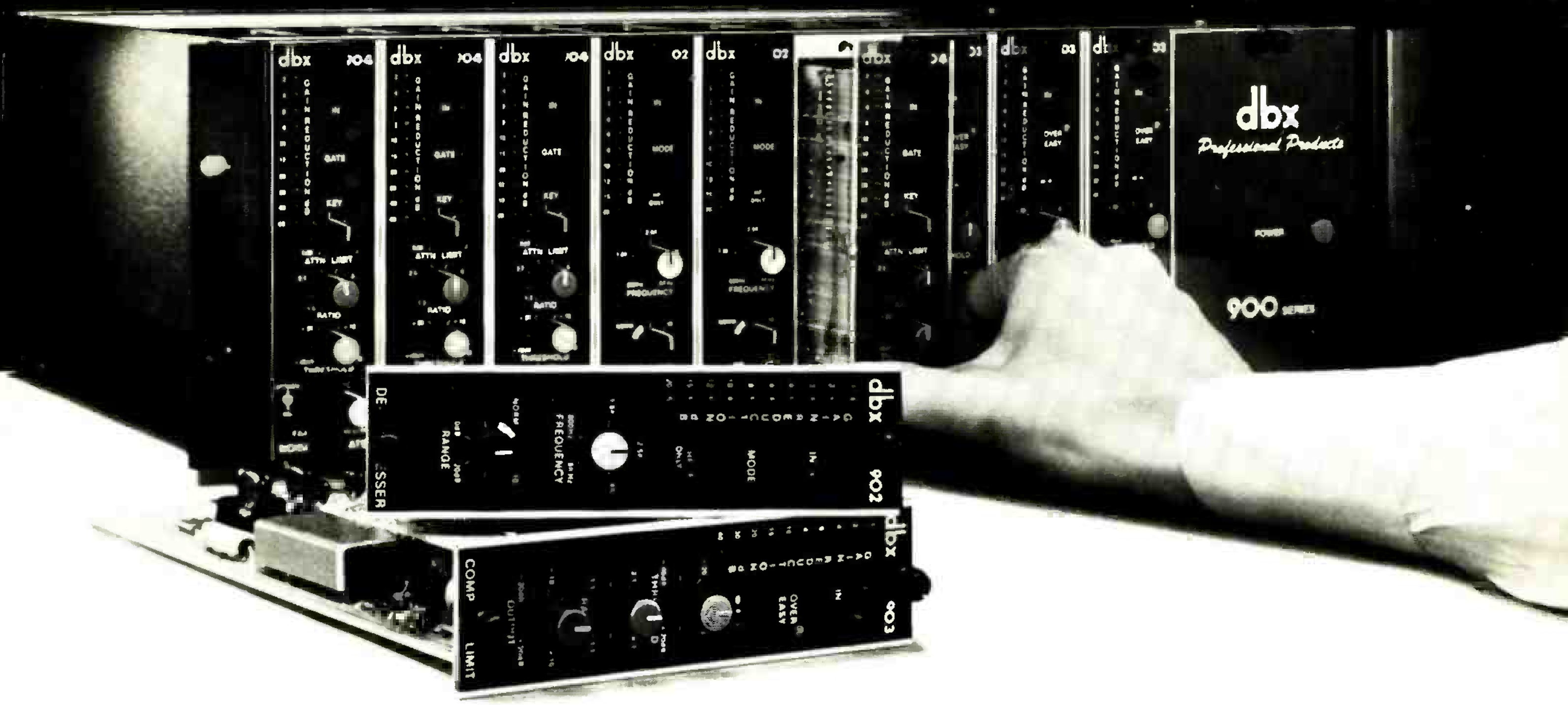
Up to eight sophisticated signal processing modules can be fitted into a rack mount unit measuring just 5¼" high. Designed for fast installation, standard connectors enable the rack to be wired easily into a system. The interchangeable signal processing modules slip in and out in seconds and offer the user both unique expression and ultimate flexibility in sound production.

The dbx Model 902 De-esser can be used as both a conventional broadband de-esser or for attenuating only a user-determined portion of the high frequency range of the audio signal. Gain reduction is adjustable from 0 to 20 dB. A unique feature of the 902 is its continuous analysis of the input signal spectrum which provides the exact amount of de-essing selected, regardless of signal level. Because the 902 does not require recalibration for signal level changes, the user can just set it and forget it.

The dbx Model 903 Compressor offers a unique new compression feature that begins reducing output volume once the threshold is exceeded. Signals are given a new and unusual sense of "punch." Like the acclaimed dbx 165

THE NEW 900 SERIES MODULAR SIGNAL PROCESSING SYSTEM.

IT'S COMPACT. IT'S FLEXIBLE. IT'S dbx.



With rack space at a premium, you want to save space anywhere you can. That's why dbx is introducing the new 900 Series Modular Signal Processing System.

You start with a single, easy-to-install 5¼" x 19" rack with built-in power supply. Then just slip in the modules you need. Up to 8 dbx signal processors, with storage for a ninth.

But the modules themselves are the real stars.

Our Model 902 is the only de-esser that continuously analyzes the input signal spectrum, providing the exact amount of de-essing you want regardless of signal level. And the 902 can be used broadband or on high frequencies only.

The 903 Compressor offers a special negative compression feature. In use, it actually begins to attentuate at the threshold, which gives the signal a new sense of punch. Of course the 903 also features our Over Easy compression as well as true RMS level detection.

Our 904 Noise Gate features adjustable attack and release rates, Over Easy downward expansion, a special key input that allows you to gate one instrument by another, and a unique "gate" mode which eliminates the need to gain ride solos during multi-track mixdown.

The 900 frame accommodates dbx noise reduction modules as well.

And this is just the beginning of our signal-processing system. Soon we'll be offering an equalizer, a flanger, and more.

So now you've got a signal processing system that's everything you want.

It's compact. It's flexible. Best of all, it's dbx.

dbx, Incorporated, 71 Chapel St., Newton, MA 02195. 617/964-3210.

dbx®

for additional information circle no. 63

New Products

Compressor, the new 902 is an Over Easy® compressor. It offers a soft-knee threshold that increases compression ratio gradually over a range of several dB. It features true RMS level detection, continuously variable compression ratios, and a threshold that is adjustable from -40 dB to +20 dB.

The dbx Model 904 Noise Gate is reportedly the ultimate noise gate, with a combination of features not found on any other noise gate, at any price. It features adjustable attack and release rates, threshold adjustment from -30 to +10 dB, attenuation limit adjustment from 0 to 60 dB, with dbx Over Easy® downward expansion for a smooth sound. It also features a KEY input that allows gating of one instrument by another.

The special GATE mode of the 904 allows users without automated consoles to put threshold programmed muting on solo channels. After the user sets the correct solo level on the console, the 904 will automatically keep the channel muted, eliminating spurious signals which frequently precede the solo itself. When the solo begins, the 904 will un-mute the channel, allowing the solo into the mix at the pre-set level.

Additional modules are under development.

dbx, Incorporated
PROFESSIONAL PRODUCTS DIVISION
71 CHAPEL STREET
NEWTON, MA 02195

for additional information circle no. 64

EECO SMPTE/EBU TIME CODE READER

Featuring exclusive tach-pulse operation, this new EECO product reads standard SMPTE/EBU edit code used for electronic indexing of video audio tapes.

Within the microprocessor-based system, time code data is verified and processed "on time." Each valid time code frame is updated prior to being outputted to ensure correct output time data associated with the reference frame pulse.

Time code input circuits read at rates from 1/16 to 60 times play speeds. If tach pulses are used, the reader automatically will switch to tach mode when code is invalid at both high wind speed and below normal play speed. Code frame rates of 24, 25, and 30 frames per second are detected automatically within the reader.

Other operating features include: restored serial code output for code dubbing; hexadecimal display for full binary word display; time code and/or tach pulse operation; optional synthesize serial code output when operating from tach pulse mode only.

The reader, EECO model TCR-650, is priced at $2,490. Synthesize serial code output option is $250. Delivery ranges from 60 to 90 days after receipt of order.

EECO, INCORPORATED
1601 EAST CHESTNUT AVENUE
SANTA ANA, CA 92701
(714) 835-6000
(No Bingo)

LEXICON ANNOUNCES LOW COST DIGITAL DELAY FOR SMALL STUDIOS AND ENTERTAINERS

The PCM 41 (Baby Prime Time) is based on the technology developed for larger systems and has exceptionally clear audio performance. It employs studio quality pulse code modulation (PCM) encoding for all delayed audio signals. Bandwidth is 20 Hz to 16 kHz with less than 0.1% distortion at all frequencies and delay settings.

The PCM 41 provides entertainers with a full repertoire of creative musical effects including double tracking, flanging, vibrato/tremolo, arpeggio, doppler pitch shift, slap echo, infinite repeat, etc. An envelope follower control

provides an articulated sweep for dramatic musical effects. An infinite repeat control is provided allowing an audio segment to be repeated indefinitely without audio degradation.

When used in conjunction with extended delay control a musical segment up to 800 ms long can be locked up and repeated indefinitely for as long as the musician wants it for a background rhythm or counterpoint.

Lexicon's PCM 41 features convenient human engineered controls for on-stage use by busy musicians and/or engineers. Also, major functions can be foot switch controlled. The system contains 400 ms of delay in X1 Mode (full bandwidth) and 800 ms in X2 Mode. Suggested retail price is $1,095.

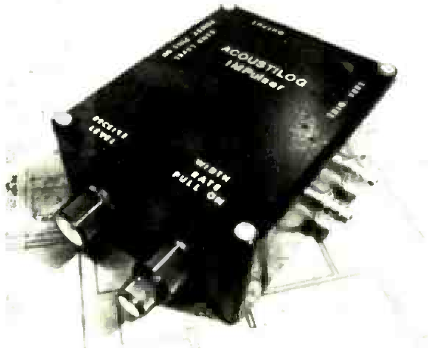
LEXICON, INC.
60 TURNER STREET
WALTHAM, MA 02154
(617) 891-6790
TELEX: 923468

for additional information circle no. 66

ACOUSTILOG, INC., ANNOUNCES NEW LOW COST IMPULSE MEASUREMENT SYSTEM

The Acoustilog IMPulser is a high performance acoustical measurement system that visually displays loudspeaker polarity, phase alignment, time delay, and many other acoustical phenomena on any triggered oscilloscope. A marked improvement over "red light/green light" polarity testers, the IMPulser features low cost, higher accuracy, and greater versatility. It is said to be indispenable for any sound installer or acoustical consultant.

Among the other important features of the Acoustilog IMPulser are:



Variable frequency - 40 Hz to 10 kHz; variable repetition rate - .3 to 10 pulses/second; variable send level; variable receive level; pause switch; AKG phantom powering; detects speaker clipping; ringing; flutter echo; rear wall slap echo, etc., and battery powered.

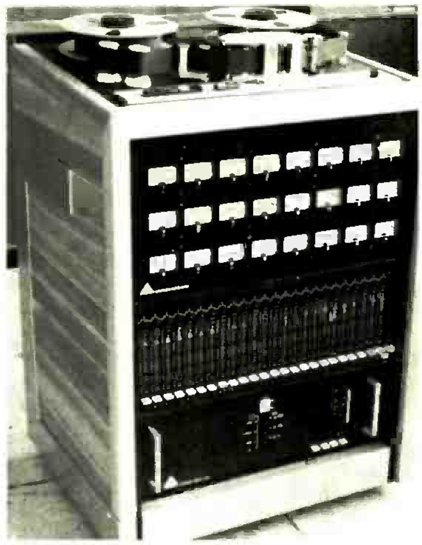
The unit is available as a stand-alone or as an option to the 232A Reverberation Timer.

ACOUSTILOG, INC.
19 MERCER STREET
NEW YORK, NY 10013
(212) 925-1365

for additional information circle no. 67

NEW TRIDENT T.S.R. MULTITRACK RECORDER

Trident Audio shocked and surprised many people at the APRS Show (London) by announcing and showing for the first time their T.S.R. multitrack tape machine.



New Products

Developed from ideas discussed at meetings with Trident studios personnel, the new recorder includes some unique features such as single button SYNC/SAFE to SYNC/READY control to pre-selected channels.

Tape speed indication is on the remote, as is a coarse and fine varispeed adjustment, from 6 ips to 45 ips. It is an extremely compact remote measuring only 30" wide by 25" deep by 45" high, including spools. The deck is 2" solid aluminum casting, angled slightly, and has 14" spool capacity.

The autolocate system is the highly successful and proven XT 24 series by Audio Kinetics which offers full memory functions and return to zero.

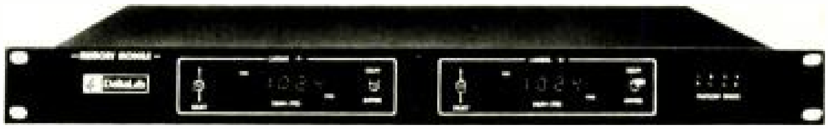
Amplification is plug-in one-card modules and features separate high frequency and low frequency equalization adjustment from SYNC and REPRO on all speeds. Inputs are electronically balanced and transformers are used only on the record and replay heads.

Common capstan frequency of 9.6 kHz ensures that the T.S.R. will interlock with other tape machines for 48-track recording.

The overall finish of the machine is natural English Ash and is styled in the same lines as the T.S.M. and Series 80 consoles.

TRIDENT AUDIO DEVELOPMENTS
POST NO. 38 - STUDIOS ROAD
SHEPPERTON, MIDDLESEX
TW17 0QD ENGLAND
TELEPHONE: (09328) CHERTSEY 60241
TELEX: 8813982 TRIMIX G.

for additional information circle no. 69



DELTA LAB RESEARCH "MEMORY MODULE"

This new product is a companion product to the highly successful DL-2 Acousticomputer® and its newest product DL-4 Time Line.™ The Memory Module is currently being delivered.

The Memory Module when interfaced with the DL-2 Acousticomputer® and the DL-4 Time Line™ allows the user an additional two full seconds of delay, with — according to the manufacturer — no degradation in performance.

The Memory Modules can be cascaded to obtain additional seconds of delay still without any degradation in performance of the master unit or the Memory Module.

DELTA LAB RESEARCH, INC.
27 INDUSTRIAL AVENUE
CHELMSFORD, MA 01824
(617) 256-9034

for additional information circle no. 70

SONY'S AFFORDABLE ECM-989 PROVIDES HIGH PERFORMANCE M-S STEREO MIKING

A unique new design makes this a very special M-S microphone. Conventional M-S mikes use a cardioid capsule and a bi-directional capsule to achieve the stereo pattern. Optimally, both capsules should be matched for frequency characteristics, however this is difficult to achieve. For ideal response the ECM-989 uses three identical cardioid capsules — one mid capsule with front orientation and two opposed side oriented capsules.

The ECM-989 utilizes gold evaporated 6u polyester film diaphragms contributing to excellent transient response characteristics. A frequency response of 20 - 20,000 Hz and signal-

to-noise ratio of 66 dB further attest to the performance capabilities of the microphone.

The docking two-part design permits the ECM-989 to adapt easily to remote control operation. The pickup qualities can be varied from mono to 150° stereo by adjusting the gain of the S capsules relative to the M capsule using the directivity control on the power supply unit. The balanced, low impedance design allows remoteability at distances of up to 100 meters without a rolloff in high frequency response. An accessory 10 meter cable is available for remoting the capsule units.

The control unit of the ECM-989 has a fully adjustable directivity control that can be continuously varied from 0° to 150° with a detent provided at 120°. A low-cut switch is provided to allow an appropriate bass rolloff for close miking without proximity effect. Rolling off at 150 Hz, the response is down 20 dB at 30 Hz. The power supply contained in the control unit requires a 1.5 volt AA-type battery that provides long life operation due to low current draw. A DC-to-DC converter raises the battery output to 9 volts, high enough to operate the circuit and deliver a wide dynamic range. A battery check LED lamp flashes when the microphone is switched on.

Supplied accessories include a 2.5 meter output cable with left and right XLR-3 -12C connectors, stand adaptor, and urethane windscreen.

Suggested retail price of the ECM-989 is $435.00. The EX 10CS-5P 10 meter remoting cable is available for $40.00.

SONY INDUSTRIES
9 WEST 57TH STREET
NEW YORK, NY 10019

for additional information circle no. 72

INOVONICS INTRODUCES TWO NEW RECORDING ELECTRONICS

Introduced as the Models 370 and 380 magnetic recording electronics they are fully self-contained units that replace existing electronics in professional audio recorders. The 370 and 380 represent the fourth generation of recording electronics made and sold by Inovonics.

The Inovonics 370 is designed primarily for those who want substantially better sound and reliability than their present recorders offer without having to invest a large sum of money in new, expensive equipment. It features low-noise circuitry, and is compatible with most studio transports and a wide variety of original and replacement heads at a cost of $580.00.

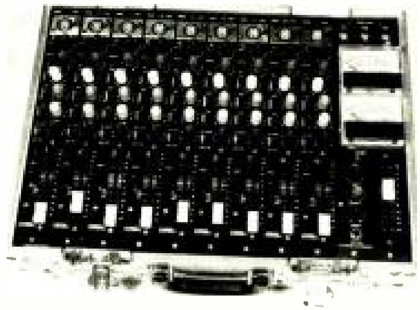
The 380, which costs $820.00, is the upgraded successor to Inovonics' Model 375, used in recording studios and radio stations around the world. The 380 can be used to create brand-new, high performance recording equipment, or to improve the performance and reliability of old tape or mag-film recorders. Its features include: special circuitry to reduce the effects of tape compression and phase distortions; superior signal and bias headroom for full compatibility with highest-coercivity tapes; standard EQ and bias settings, as well as an optimized mode with separate setup for best performance from "super" tapes; SYNC, auto-mute, and remote control of all functions.

INOVONICS, INC.
503-B VANDELL WAY
CAMPBELL, CA 95008
(408) 374-8300

for additional information circle no. 73

INTERFACE 200 - 8X2 BATTERY POWERED MIXER

The Series 200-8x2 is intended for film and TV use and all mono or stereo recording or mixing applications requiring a small battery-operated fully professional mixer. It is built for ruggedness and portability in a 13" x 17.5" x 7" (with lid, 4" high without lid) ATA-style case with handle, and weighs 22 pounds with the lid.



The new 8x2 mixer will operate about 20 hours from an external rechargeable Gel-Cell battery pack (supplied) or on any 12 volt source such as a car battery, or an optional AC supply. The model in the photo shows 9 inputs, but it can optimally be supplied with 8 inputs and a

communications module providing for intercom, slating, playback, and test tones. Inputs include phantom power, phase reverse, cue and echo sends, panpot, 4-position input pad before transformer, 4-position input gain set switch, three equalizers with 4-position mid frequency select, 4-position low cutoff, ±6 dB gain trim, solo, on/off/mute switch (module draws no current when off), and Duncan professional conductive plastic slider with dust seal. Two standard VU meters, ganged Duncan master, and phones monitor are on master panel.

INTERFACE ELECTRONICS
3810 WESTHEIMER
HOUSTON, TX 77027
(713) 626-1190

for additional information circle no. 75

SPECK MODEL "800-D" MIXING CONSOLE

The Speck 800-D is a 28 input, 16/28 output studio mixing console. The console is totally modular with 28 input modules, a master module, and a complete communications module housed in a sturdy mainframe that contains 16 large illuminated VU meters.

Each input has 8 pannable assign, 3-band parametric equalizers, 3 sends, pan, stereo solo, a long throw slide fader, and most important: a second line input with an independent slide fader, a 2-band equalizer, and pan.

Since the Speck 800-D has two (2) discrete line input circuits for each input module and 28 assignable direct outputs in addition to the 8 submasters, the 800-D is well suited for 16-, 24-, or 32-track studio operations. The stereo program bus is independent of the multitrack assign section which allows the console to feed a full compliment of ½-track, ¼-track, and cassette recorders simultaneously during mixdown.

A 384 point patchbay is standard on all "D" models, and is wired to accept two (2) 16-track, 24-track, or 32-track tape recorders. All connections for tape recorders, power amps, chambers, and outboard equipment are made via eight high density multi-pin connectors at the rear of the console.

Model SP-800-D-28 is priced at $25,190.00.

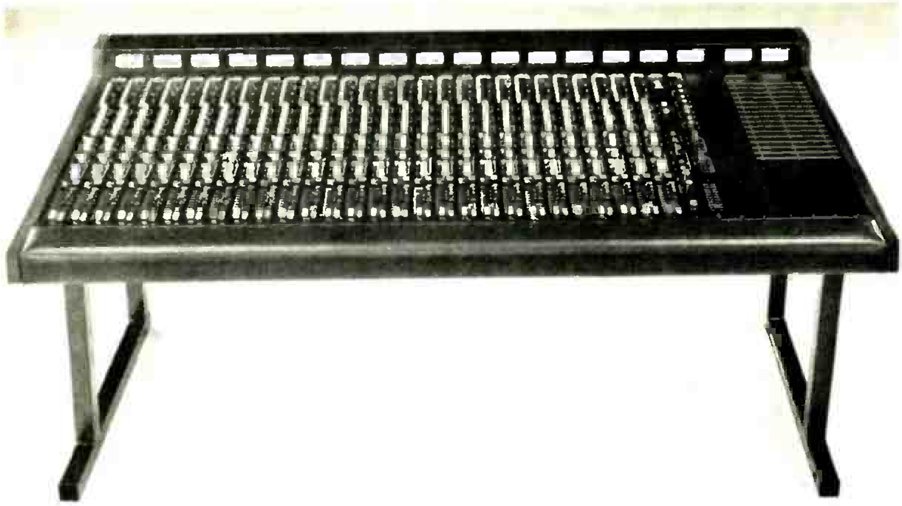
SPECK ELECTRONICS
7400 GREENBUSH AVENUE
NORTH HOLLYWOOD, CA 91605
(213) 764-1200

for additional information circle no. 76

SONTEC FOUR-CHANNEL, THREE-BAND MASTERING EQUALIZER

Introduced as the 430B Parametric Disk Mastering Equalizer the unit uses four channels of electronics to permit the mastering engineer to equalize both the preview and program channels simultaneously. Contemporary cutting systems require this simultaneous equalization to enable the cutting system computer to maintain continuous pitch and depth control for optimum conservation of available groove area.

Each of the four equalization sections of the 430B consists of a three-band shelving parametric equalizer with boost or cut of up to 12 dB in 1 dB increments within the overlapping ranges of 11 to 570 Hz; 120 to 6,800 Hz; and



3,400 to 25,600 Hz. Slope or "Q" can be varied from 5 to 15 dB/octave while maintaining constant amplitude. The high shelf provides up to 12 dB boost or cut in 1 dB increments, at 10 kHz and the low shelf at either 50 or 100 Hz.

Usable dynamic range of the equalizer is 110 dB and crosstalk between any two channels is at least 90 dB down. A proprietary direct-coupled discrete operational amplifier with slew rate of over 200 volts per microsecond is used throughout the signal path.

The Sontec parametric mastering equalizer offers a number of in-use advantages to the production mastering house. Eleven controls are clearly marked and detented to make it possible to reproduce all settings to a high degree of accuracy to precisely duplicate original lacquers, or to produce identical "clone" lacquers, at different plant locations.

The 430B has 21 dB of headroom above +4 dBm to allow adequate equalization flexibility for tape masters recorded at tape saturation level to be reproduce at normal levels. The equalization function can be controlled by the banding logic in the cutting system. The equalization frequencies are chosen to allow the full range of the system to be used in half-speed cutting operations. Two Sontec 430B equalizers may be hooked in tandem and switched during banding to provide completely different equalization characteristics for successive bands.

The equalizer is supplied in a standard seven-inch rack mount and is available for use with all international line voltages. The 430B carries a two-year warranty covering both parts and labor.

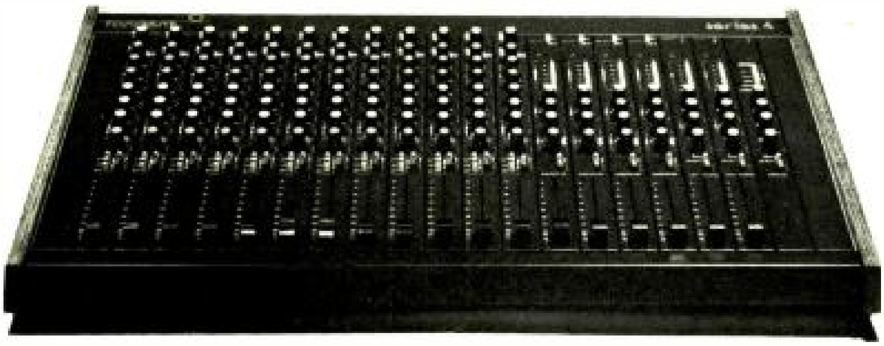
SONTEC ELECTRONICS
10120 MARBLE COURT
COCKEYSVILLE, MD 21030
(301) 628-2283

for additional information circle no. 78

ELECTRO-VOICE INTRODUCES NEW MIKES FOR THE '80s

The new dynamic cardioid vocal microphone designated the PL80 was designed specifically for the professional vocalist. The PL80, it is claimed, represents the successful culmination of the application of a totally new concept in computerized microphone design — a concept that allows E-V engineers to predict precisely how a microphone will sound in a "live" environment while it is still in various design stages.

The application of this design concept called "fast Fourier transform" (FFT) is claimed to be another E-V inovation in the field of sound reproduction. In addition, extensive field tests utilizing top performing vocalists such as Steve Perry, of Journey, as well as interviews with some of the best sound men in the field were conducted to determine exactly what they needed in a vocal mike. The new PL80 is said to give the vocalist the kind of sound he or she is asking for · a mike that actually enhances the voice without compromising individual quality. The unit is claimed to excel in important areas like gain-before-feedback



and sensitivity.

The PL80 features a shock mount to reduce handling noise plus a built-in Acoustifoam™ blast filter to reduce *P-popping*. Designed to handle rough treatment, the PL80 is made of aluminum and diecast zinc, and uses a dent-resistant Memaflex grille screen.
Suggested retail price is $199.95.

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

for additional information circle no. 80

TANGENT SERIES 4 MIXING CONSOLES

Joining their stereo bus "AX" series, the new Series 4 (four output bus) console is designed to offer the professional the optimum features and performance for both sound reinforcement and 4- or 8-track recording sessions. Offered in either 12 input or 20 input fully modularized mainframe, some of its features are:

Transformerless input circuitry; three-band, continuously variable equalization in each channel; peak LED and 20 dB pad on each input; eight independent returns, PFL, and six-out bus assign through submasters (4) and R/L stereo busses; three independent foldback sends; full provision for multitrack monitoring and assign; muting, 100 mm faders, phantom mike power; external power supply, and many options including reverb and expander modules.

The Series 4 is now shipping from stock and is priced from $2,800 (professional suggested net price).

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news

WESTLAKE AUDIO PURCHASES SECOND DIGITAL MULTITRACK SYSTEM

Westlake Audio has purchased and received a second 3M digital multitrack system, consisting of four-track and 32-track recorders, only nine months after receiving its first.

"A second system gives us, and our clients, much more flexibility in scheduling digital projects – from start to finish," according to Glenn Phoenix, Westlake's president. "It also allows more artists to take advantage of the multitrack digital equipment.

"As the end-product (the digitally mastered disc) gets more into the market, digital is becoming more of a reality for my clients," says Phoenix. In addition to those recording artists willing to pioneer new technology, more and more Westlake clients have become interested in digital after hearing it.

Phoenix also reports an interest in the recording industry for digital rentals. Besides being moved between Westlake's two studios, the 3M system has been rented to producer Val Garay at Record One for artists such as Linda Rondstadt and Waddy Watchtel. Says Garay, "digital is inevitable because the quality of the end-result is so much better than analog."

Digital recorders have been also rented by Doug Sax, of The Mastering Lab, and by Nonesuch Records to record the "Silverlake" opera at the CBS studios in New York City.

Use of the multitrack system at Westlake studios has included other Nonesuch recordings and the Australian Group, Angel City. Future uses include producer Giorgio Moroder who reserved the system starting in July for a forthcoming Donna Summer album.

Westlake also offers 3M's electronic digital editing system and a digital preview unit. The preview unit retains and delays the signal in the digital domain up to the record-cutting lathe. According to Phoenix, "precise and previewable electronic digital editing of up to 32 tracks, simultaneously, brings digital recording into the forefront of state-of-the-art studio technology."

SRA HOLDS THREE-DAY MUSIC FESTIVAL

The Southern Recording Association met on August 8, 9, and 10th for its 1980 SRA Music Conference. The association, whose membership is made up of Orlando area recording studios, hosted the three-day affair feature 17 guest speakers including Ed Shea (ASCAP), Jerry Smith (BMI), Dianne Petty (SESAC), and Nancy McAleen (U.S. Copyright Office). Also included in this year's line-up are Eric Schabacker (Bee Jay Recording Studios), Bob Todrank (Valley Audio) and local attorneys Jay Willingham and Herbert Allen.

In addition to the three seminars and six workshops covering such subjects as copyrighting, publishing, recording contracts, sound reinforcement, legalities, careers in recording, and songwriting techniques, the event offered its participants admission to an ASCAP sponsored cocktail party, a tour of SRA member studios, and a Certificate of Completion.

This year's conference was the third SRA sponsored event thus far.

HEIDER TO RETURN TO WALLY HEIDER RECORDING, OTHER FILMWAYS ASSIGNMENTS ANNOUNCED

Dave Kelsey, President of the Filmways Audio Group, announced that Wally Heider will return to the studios which bear his name, where he will assume the position of Director of Operations, effective September 15. Heider, who left the studio in 1973, is recognized as being one of the early innovators in the field of remote recording.

In an additional announcement, Kelsey named Peter Butt as Chief Engineer and Director of Maintenance at Heider Recording, and Bill Isenberg as Chief Engineer of the Filmways Audio Services complex in the San Fernando Valley.

B&B AUDIO ANNOUNCES REORGANIZATION AND MOVE

B&B's principals now include David Baskind, E. J. Bissot, and Bill Kaufman in the role of general partners, and R. Swettenham, founder of Helios Electronics, as a limited partner.

In addition, B&B is no longer affiliated with Aphex Systems, and is engaged in setting up its own worldwide rep network.

B&B was founded in 1972 and specialized in refurbishment, studio acoustics, and custom console design and construction until its affiliation with Aphex in 1978.

The company is again engaged in these activities and will also be offering mobile system design services featuring Swettenham's extensive experience. Their new address is: 6762 Las Olas Way, Malibu, California 90265. Telephone: (213) 461-2572.

SCHARFF COMMUNICATIONS EXPANDS PRO-SOUND DEPARTMENT FOR RECORDING STUDIO INDUSTRY

SCI, the ten-month old professional sound rental and sales company located at 1600 Broadway, in New York City, has announced expansion of its services for the recording studio industry. The expansion features an exclusive rental relationship with Empirical Audio, of Ossining, New York, in the film, video, and recording fields.

The joint venture, announced by SCI president Peter B. Scharff and Win Schwartau, of Empirical Audio, will give Empirical Audio higher visibility in New York City, including a demo room at the SCI facility. The arrangement will also add to SCI's already extensive inventory of professional audio equipment. The new lines represented with Empirical Audio will include Advanced Music Systems, Ltd., Audio Kinetics, Ltd., Trident Audio Developments, Ltd., Valley People, Inc. (formerly Allison

— *continued overleaf . . .*

BLONDIE RECEIVES 150th AMPEX GOLDEN REEL AWARD

The rock group, "Blondie," has become the 150th winner of the Ampex Golden Reel Award for the success of their gold album, "Eat To The Beat."

Presenting the latest of their four Golden Reel Awards to Blondie were Richard Antonio, national sales manager and Peter Cain, consumer market development manager of the Magnetic Tape Division of Ampex Corporation.

The organization benefiting from Blondie's Golden Reel Award is the Police Benevolent Association of New York City. Receiving the contribution from Blondie and Ampex was Robert DeVito, financial secretary for the P.B.A..

In addition to the group members of Blondie, "Eat To The Beat's" producer Mike Chapman, the recording engineer for the album, Dave Tickle, assistant engineer James Farber, and the Power Station in New York City, the studio where the album was mastered, received Golden Reel Awards as well.

Blondie will continue being featured in Ampex advertising with a lineup of other top recording artists who use and endorse Grand Master™ recording tape.

Research), Eventide Clockworks, and Dolby Laboratories, Inc., among others.

SCI also announced the inclusion of two additional lines under its representation for sale and rental: Orange County Electronics and the full line of RTS Systems.

SCI president Peter B. Scharff was formerly associated with WNET-TV and was Associate Producer of the Emmy Award winning "Live from Lincoln Center" series. Since its founding last August, Scharff Communications has been growing steadily. "Our growth has been five or six times greater than I expected when we started," says Scharff. "Every month we have have an increased inventory, billings, and staff." The firm supplies professional sound equipment to the film, video, and recording studio market.

SINE QUA NON (SQN), CONCORD JAZZ, VARESE SARABANDE & CHALFONT RECORDINGS RELEASED ON dbx® ENCODED DISC FORMAT

The first digital recording by Sine Qua Non is being released as a dbx Encoded Disc. The album, *Digital Hits of 1740*, includes works by Pachelbel, Albinoni, J. S. Bach, Handel, Corelli, and Mouret. Performances are by the Cambridge Chamber Orchestra which is composed of members of the Boston

Symphony Orchestra and Empire Brass Quintet.

According to SQN president, Joan Grow, "The dbx Encoded Disc format allows the full dynamics of the Soundstream digital recording to be enjoyed without annoying surface noise. We look forward to issuing more of our albums as dbx discs."

Three albums by Morton Gould conducting the London Symphony Orchestra are being released on the Varese Sarabande and Chalfont labels as Digital dbx Discs. These are the initial issues in a series of digital recordings in the dbx Encoded Disc format to be made available under these labels.

The three London Symphony Orchestra releases — the first digital recordings made by the orchestra — use the Soundstream digital recording system.

Two of the three albums are on the Varese Sarabande label: *Latin American Symphonette* (a collection of original compositions by Morton Gould), and *Digital Space* (a concert of symphonic film music by American and British composers). The third album which is on the Chalfont label features orchestral showpieces including: Ravel's *Bolero*, Ginastera's *Estancia Ballet Suite*, and Weinberger's *Polka and Fugue from Schwanda*. The three albums were produced by Jerome E. Ruzicka, dbx vice president and director of the dbx Encoded Disc Program.

"Since the conventionally pressed versions were released last year, each of these albums has been well received in the audiophile record market," said Mr. Ruzicka. "Now, their quality will be further enhanced by the dbx Encoded Disc format since the full dynamics of the original live performance is reproduced without the annoyance and distraction of record surface noise."

According to Varese Sarabande vice president Tom Null, "dbx is what all the noise *isn't* about — pure music." In his statement, Chalfont president Tom Britton said, "We are delighted that our recording of *Bolero* will now be available as a Digital dbx Disc. For the first time, the listener can experience the full dynamic range of this popular work from the quiet beginning passage to the thunderous finale."

The dbx Disc versions of the Varese Sarabande and Chalfont albums by the London Symphonic Orchestra were mastered by Bruce Leek at IAM Studios, in Irvine, California.

These new additions to the Digital dbx Disc Library, which require an inexpensive decoder for proper playback, will be available through the national network of dbx retailers.

Selected titles from the Concord Jazz Records catalog will be issued as dbx Encoded Discs. The initial releases on the Concord Jazz label will include recent albums by Laurindo Almeida, the LA-4, and Cal Tjader, according to Carl E. Jefferson, president of Concord Jazz.

"We have always prided ourselves in bringing the best jazz artists to the public with recordings of high quality. Participation in the dbx Encoded Disc Program represents another step we are taking to present our artists in the best possible light to jazz enthusiasts around the world," said Mr. Jefferson.

news

SOUNDESIGN/TELAUDIO ANNOUNCES DESIGN SERVICE

Oliver Berliner, who has gained note as a leading designer of teleproduction studios as well as a major distributor of many important video products, has announced that his design services are now available to recording studios.

Recognizing that the video revolution is here and that leading recording studios will need to interface their audio with the video product being made available to the home viewer, and for the emerging pay TV market, Berliner, an internationally known author of some twenty-dozen articles and technical papers on music, audio, and video states: "The service we can offer makes any recording studio fully video operational without its staff having to attend far-away seminars or spend hundreds of hours in video familiarization."

Berliner's SounDesign Engineers will cater a video capability to the studio owner's desires, and his Telaudio Centre will provide all the necessary equipment if the recording studio owner has no favorite vendor from whom he wishes to buy it.

SOUNDESIGN/TELAUDIO is at Dept. R, P. O. Box 921, Beverly Hills, CA 90213.

SONY DIGITAL TO BE USED ON SPRINGSTEEN PROJECT

A new record project by CBS artist Bruce Springsteen was recently recorded and is currently being edited with Sony digital audio equipment. Springsteen and producer Jon Landau are now working on the project at Clover Studios, in Los Angeles.

The equipment used includes the Sony PCM-1600 digital processor, as well as a prototype digital editor and BVU-200As for the master tape.

According to Dan Morehouse, of Clover, "After side-by-side comparison of the Sony digital system with our own analog system, everyone here enthusiastically chose the Sony PCM-1600."

Clover is using the services of Digital Sound Recording, a new digital recording service company that has been in operation since January. Van Webster, owner of D.S.R., reports that the services include a full range of digital recording and editing. "We provide the equipment and technicians to handle all aspects of digital recording, so that the artists are free to concentrate on the aesthetics of their music," he said. For the Springsteen project, technician Jim Bauerlein is operating the Sony system.

JOURNEY RECEIVES AWARD FROM ELECTRO-VOICE

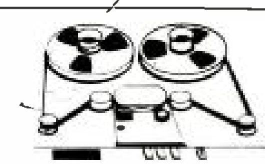
A plaque featuring a gold-plated PL80 microphone was presented to Steve Perry and the rock supergroup Journey during a recent tour stop at the University of Notre Dame near E-V's corporate headquarters in Buchanan, Michigan. The award was presented by Chuck Gring, E-V's Music Products Sales Manager. The plaque reads "In recognition of their support and contribution to performing excellence Electro-Voice presents Steve Perry and Journey the first PL80 microphone." Journey has been associated with Electro-Voice for close to two years. During that period they have participated in joint tour promotions with E-V dealers and have been an invaluable source for field testing of new E-V music products.

Perry's gold-plated PL80 was the first production unit of this new vocal mike to come out of E-V's manufacturing facility, and was plated and appropriately engraved with Perry's name and the mike's serial number "00001."

The PL80 was designed with a computer-assisted technology called "fast Fourier transform" (FFT) which allows the design engineer to precisely predict exactly how a microphone will sound in actual use — not just in sterile engineering test environments, E-V's newest mike enhances the voice without compromising individual vocal quality.

PHILIPS, SONY PROPOSE OPTICAL DIGITAL AUDIO DISC STANDARD

N.V. Philips Gloeilampenfabrieken of The Netherlands and Sony Corporation of Japan announced together that their mutual cooperation has led to further improvements in the optical DIGITAL COMPACT DISC system which was announced by Philips in March 1979. The improvements are in the areas of modulation and error correction which will permit 60 minutes of high-density recording on one side of the 12 cm (4.72 inch) disc.



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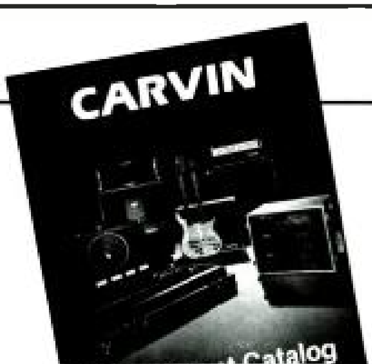

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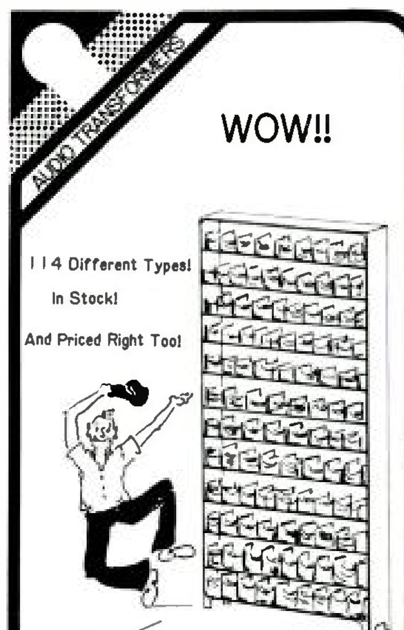

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. . . continued from page 18 —
except maybe a bit more base. You can't get involved in eye make-up and lips because that changes with the seasons and so on.

"Where men are concerned, however, where the subject ends up with dark circles and so on, you want to correct that. We suggested three shades of pancake, Tan 1, Tan 2, and Natural 2; and four pan sticks, Sun Tone, Bronze Tone, Olive, and Deep Olive. Depending on her complexion a woman can use these as well, and all can be found easily in your drugstore. I think that's as much as you really need to do. Otherwise it gets too involved."

These standard make-ups, says Salvatore, are now used regularly in television and motion pictures since film stocks have so improved that there's little need to compensate with theatrical make-up.

"Pancake is dry and mat," adds Salvatore, "while pan stick has a certain lustre to it. People could also use Erase to cover dark circles under the eyes, and I certainly would suggest a translucent loose powder to cover a shine. I've found that almost everyone looks very shiny on those cameras if they're not dusted down.

"If someone wants to be seen at their best, it would behoove them to use a professional make-up before stepping in front of the camera." The only warning here, says Salvatore, is to avoid frosted or irredescent colors which tend not to photograph very well.

The choice then is to either do a very light, basic job oneself, or to hire a professional from a local beauty salon to perform the task. He warns that a non-professional who gets too involved with make-up could simply cause his subjects to end up looking like clowns.

"Also," reminds Polon, "it's quickly apparent on camera when you're rehearsing what looks good and what doesn't. You'll know as soon as they go on camera."

In the instances of some rock acts, no make up at all may give the desired, sweaty effect that they're looking for, as in the case of the band Golden Earring when they appeared on the "Midnight Special," where their manager preferred no makeup whatsoever, only the sweat pouring off their faces under the hot lights while they performed their heavy metal repertoire.

So it is now rather evident just how involved video can be even at its most basic levels. The complexity of the medium, from the video, to the lighting, to the make-up in simply technical areas, let alone matters of business and creative judgments, should deter a number of prospective video artists at this level. On the other hand, it is no more complicated than a full-blown, Twin 24, fully automated, 56-input recording studio.

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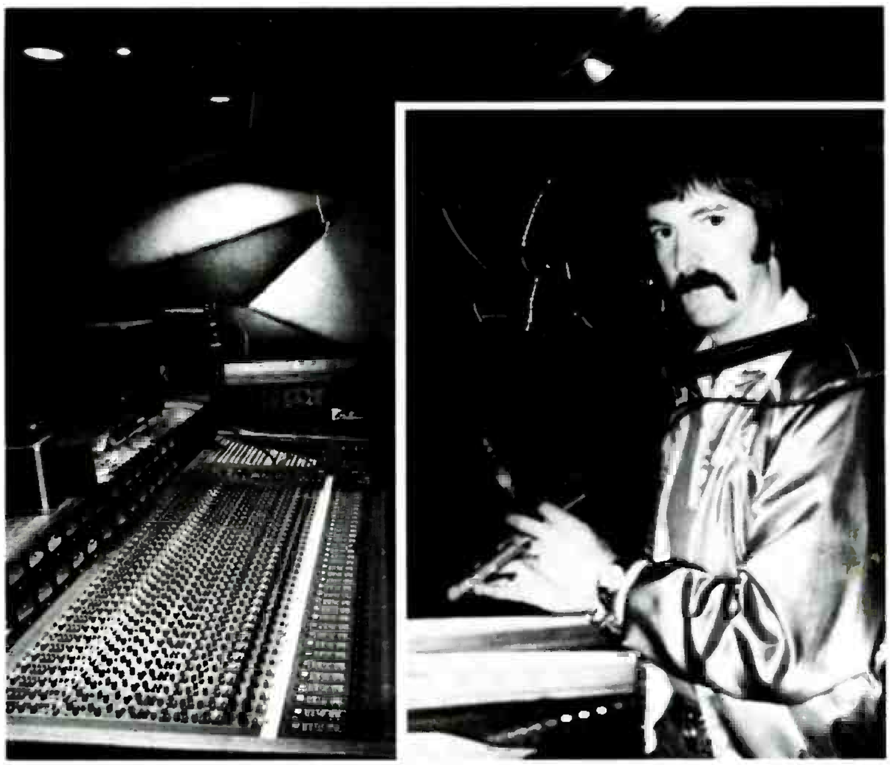
fact: "I listened to them all... and nine times out of ten, with our artists, the best microphone was the SM81"



Criteria Recording Studios,
Miami, Florida

Dennis Hetzendorfer

Dennis Hetzendorfer,
Staff Engineer

"The true sign of a really excellent microphone is that it can *maintain* its high performance, session after session after session. Here at Criteria, when the situation permits, several different microphones are set-up at each instrument, without the engineer knowing which mike is exactly where. We then fade from mike to mike and let our ears find out which is best for each application. Nine times out of ten, with our artists, the best microphone has been the SM81.

"The switchable bass rolloff and pad (a built-in 10 dB attenuator) gives the SM81 incredible versatility. We can use it with bass drums and cymbals, as well as with acoustic guitars. In fact, all the acoustic guitar segments on the Bee Gees' *Spirits Having Flown* album were recorded with the SM81.

"The SM81 really changed our minds about the ruggedness of condenser microphones. It's a precision piece of equipment, but it's durable. You don't always think about a studio microphone needing durability... after all, we don't have the rough handling problems encountered in concert recording. But, when you have a reputation as one of the most technically exacting studios in the country, you appreciate how *many* little things can subtly affect the sound of a delicate condenser microphone. The SM81 sounds good *every* time we use it... and, at Criteria, as in any good studio, we just can't afford to have a microphone we can't depend on.

"We've used the SM81 on recording sessions with the Bee Gees and Kenny Loggins and you can be sure there will be more.

"It's one great mike!"

Send for our brochure.

SM81 Cardioid Condenser Microphone by

SHURE

Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204. In Canada: A. C. Simmonds & Sons Limited.
Outside the U.S. or Canada, write to Shure Brothers Inc., Attn: Dept. J6 for information on your local Shure distributor.
Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

for additional information circle no. 104