March/April 1989
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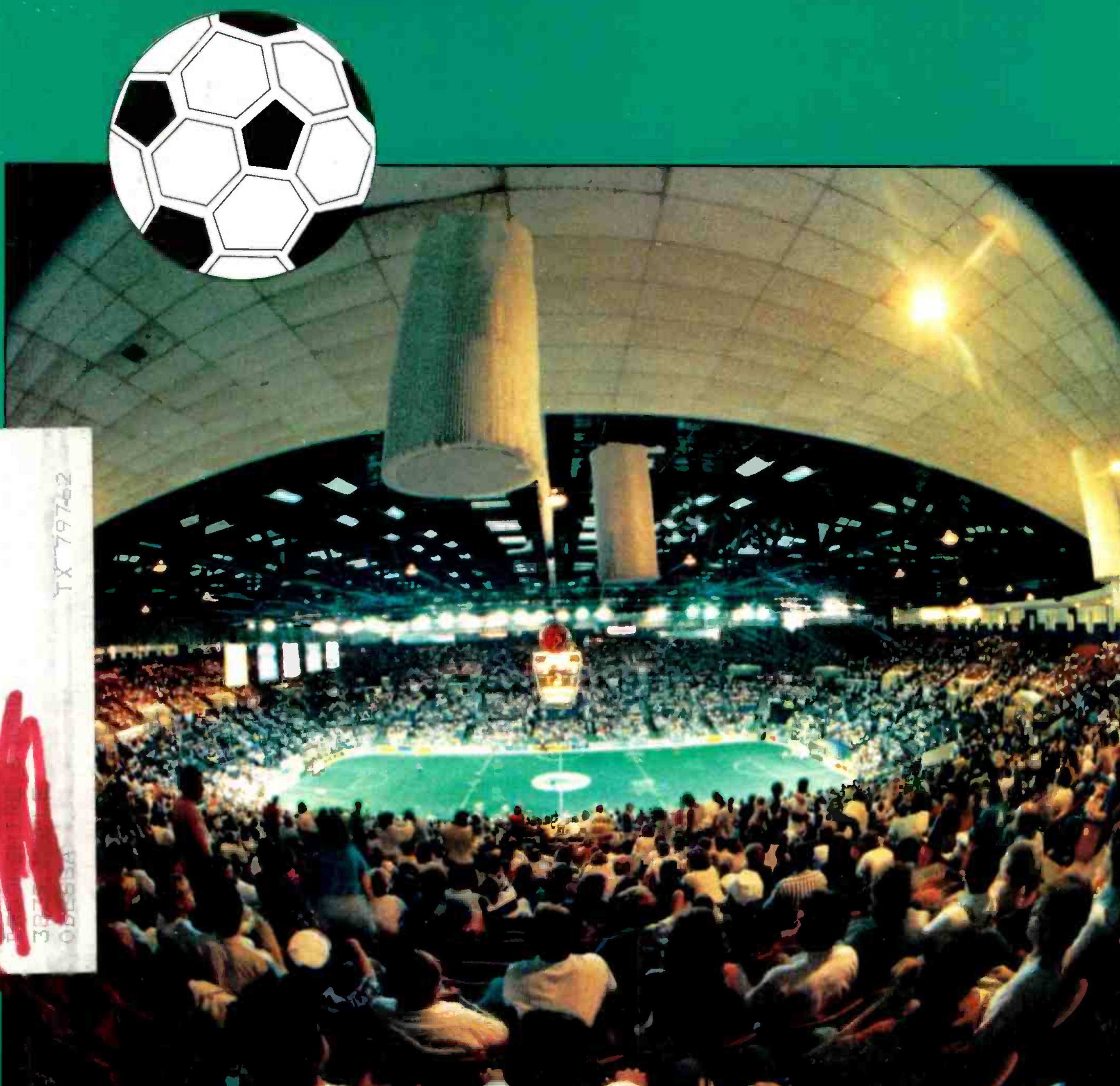

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MARCH/APRIL 1989 VOLUME 23 NO.2

the sound contracting engineer

the recording engineer

the broadcast engineer

THE ELECTRONIC COTTAGE

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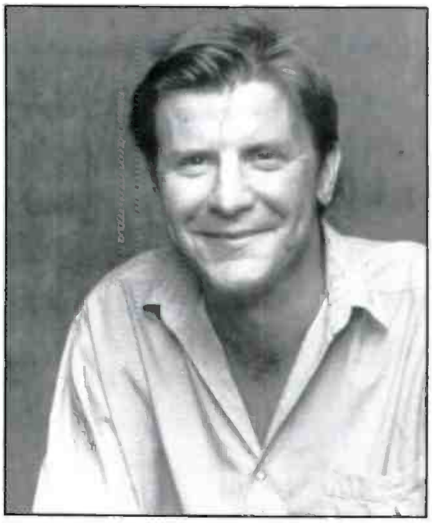

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About the Cover

• On our cover, The San Diego Sports Arena. The story begins on page 6.

Photo credit: Greg Schenewerk.

Letters

Dear Friends,

I find **db Magazine** to be very helpful and informative.

I especially appreciate the articles pertaining to sound reinforcement and small studios.

I am looking forward to my new subscription.

Also I hope that you will soon include articles on speaker placement and system design for different halls and auditoriums.

Sincerely,

Dana Allasin Powers

Messiah Sound

Lakeland, FL

Your wish is our command! See the articles in this very issue. Ed.

Dear Editor,

It has come to our attention that several comments in a recent article (see db, September/October 1988) concerning the construction of our new production facility have offended Dr. Peter D'Antonio, the founder of RPG Diffusors, and Russell Berger of the Joiner-Rose Group. In the article we related our experience with building our own studio, including the construction of custom-made diffusors, but did not credit Dr. D'Antonio for his pioneering work in designing and constructing diffusors of this type. We did not mean to make any negative inference whatsoever with regard to Dr. D'Antonio's work, for which we have a great deal of respect, nor did we mean our comments to infer that our diffusors were in any way superior to RPG diffusors. To be perfectly honest, we really did not know if our diffusors would work at all, hence the statement in the article that they "exceeded expectations." In fact, our diffusors probably do not work as well as Dr. D'Antonio's RPGs, and we make no claims whatsoever regarding their performance in respect to his. We wish to apologize to Dr. D'Antonio for the unintended slight in not crediting his design work, and deeply regret any offense caused by the article.

We did not credit Russell Berger of Joiner-Rose Group because we did not think he would want to be credited, since we did not implement all of his ideas due to cost and other considerations. Russell Berger was compensated for his advice, so his ideas were not expropriated. However, we would be the first to admit that Russell extended his help as a friend far beyond that which was requested. He noticed several mistakes and errors in calculation in our original design, and called them to our attention. In the last few paragraphs of the article, I was quoted as saying, "There were many times during the construction when I wished I had just called Russ Berger or someone, and had them do it." We have the highest regard for Russ Berger's work, and appreciate his help in designing Future Audio.

Yours truly,

Tony Rodriguez

Owner, Future Audio, Dallas

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db, The Sound Engineering Magazine(ISSN 0011-7145) is published Bi-monthly by Sagamore Publishing Company Inc. Entire contents copyright 1989 by Sagamore Publishing Company Inc., 203 Commack Road, Suite 1010, Commack, NY 11725. Telephone: (516)586-6530. db Magazine is published for individuals and firms in professional audio recording, broadcast audio-visual, sound reinforcement-contracting, consultants, video recording, film sound, etc. Application for subscription should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year($28.00 per year outside U.S. Possessions, $16.00 per year in Canada)and payable in U.S funds. Single copies are $2.95 each. Editorial, Publishing, and Sales offices are at 203 Commack Road, Suite 1010, Commack NY 11725. Second Class postage paid at Commack, NY 11725 and an additional mailing office. **Postmaster: Form 3579 should be sent to db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.**

MEYER SOUND

Sound Thoughts on Live Performance

"Creating the subjective experience of live performance sound belongs to the artists and sound designers. Our creativity comes in building speakers and systems that give the truest possible account of what the performers produce."

John Meyer, Founder and President, Meyer Sound Laboratories

John Meyer's involvement in loudspeaker design began in 1967 when, as a technician for a Berkeley, California Hi-Fi supplier, he set out to discover why a leading manufacturer's drivers kept tearing themselves to pieces. Further investigations convinced him that the market sorely needed a class of rugged professional speakers that would maintain their characteristics over time.

Research in Switzerland in the early seventies secured his knowledge base. In 1972, Meyer developed the JM3 all horn loaded tri-amp system with rigging, which was the standard for Broadway shows until the introduction of the UPA in 1980. From 1973 to 1979, Meyer sought out the best available parts and designed the first Ultra Series™ reinforcement speakers. In the decade since, John Meyer has established Meyer Sound Laboratories at the forefront of professional reinforcement technology.

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As a consequence, Meyer Sound products have earned a reputation for the highest reliability in the industry. All are guaranteed to meet or exceed specified performance levels when properly installed.

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Instead of second-guessing the tastes of the market, Meyer produces sound systems that most truly represent the character of the signal they receive, leaving artistic control where it belongs—with the artists and sound designers.

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Meyer Sound

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Calendar

• **EDS '89** (Electronic Distribution Show and Conference), the national forum and marketplace for electronic distribution, will be held at the Las Vegas Hilton Hotel, in Las Vegas, Nevada. Conferences and seminars start on Monday, May 8, and exhibits open on Tuesday, May 9 through Thursday, May 11. The annual event is sponsored by the Electronic Industries Association/Distributor Products Division (EIA/DPD), the National Electronic Distributors Association (NEDA), and the Electronics Representatives Association (ERA).

Information about EDS '89 is available from the Electronic Industry Show Corporation, 222 South Riverside Plaza, Suite 2710, Chicago. IL 60606, (312) 648-1140.

• **Pro Audio Asia '89**, the international trade exhibition for professionals in the broadcast, recording, public address, installation, contracting, and duplicating industries, will take place from July 6-8 1989 at the Hong Kong Exhibition Centre in Hong Kong.

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

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

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

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

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

• Architectural Acoustics Standards, a two-day workshop sponsored by ASTM (American Society for Testing and Materials) Standards Technology Training, will be held May 1-2, 1989 in Geneva, Illinois. The workshop will focus on the basic principles and practical applications of architectural acoustics, and will include laboratory demonstrations, a tour of Riverbank Acoustical Laboratories, and a thorough review of 15 related ASTM standards. For more information, contact Kathy Dickerson, ASTM Standards Technology Training, 1916 Race St., Philadelphia, PA 19103, (215) 299-5480.

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MICHAEL KLASCO

Acoustical Design for Large Facilities

Sound system contractors tend to view room acoustics as the dark side of audio. Certainly, all will concede room acoustics and sound system performance are intertwined, nevertheless, proposals for acoustical analysis and remedial measures are typically avoided by sound contractors. The sound system and the room are like a car on a road. The performance limits are interactive and optimum selection of one is dependent on the other. A facility with highly reflective walls restricts designing for uniformity of direct sound coverage as any spill-over will increase the perceived reverberation and may result in flutter echo. Conversely, the same space after acoustic modifications would allow a different selection of horns (a lower Q cluster) and more even coverage, as much of the spill-over would be absorbed by the acoustic treatment on the walls.

Just as equalizers, time delays, limiter/compressors are used to ensure that the sound system performs its intended functions, modification of room acoustics also are often a necessary ingredient in achieving intelligibility, non-fatiguing and aesthetically pleasing sound reinforcement. Sometimes dealing with the acoustical problems is the only way the sound system will successfully perform its intended purpose. Remember, even more than a good sound system, your client wants good sound!

While most sound contractors are not equipped to solve serious acoustical problems, without the assistance of an acoustical consultant, they can usually sense when a space has an acoustical problem. Echos and long reverberation times are not subtle defects. At times sound contractors are called in to fix sound system problems when room acoustics are the real culprit. The sound contractor can act as the early warning system when poor acoustics are encountered.

I am often contacted by sound contractors attempting to make sound systems work in unworkable spaces. Some of the sound systems have undergone component changes, re-aiming, etc., but without significant improvement. When the mid-band reverberation time (RT60) is greater than four seconds, the only electro-acoustical solutions are headphones or pew-back speakers!

During the past few years I have worked with some sound contractors in implementing acoustical solutions. Conventional surface treatments are usually installed by general contractors (or "acoustical contractors," but many of these firms are no more than dropped ceiling installers!)

In the acoustical renovations I have designed acoustic devices with wide-band, high-absorption coefficients are installed by a sound contractor that is competent with rigging techniques. Often the sound contractor may already be handling the equipment maintenance and is now being asked by the facilities management how to improve the sound quality, but other reasons point to the sound contractor installing the acoustical devices. As the sound system design is closely linked to the acoustical treatment, the sound contractor should be involved with these aspects of the job. If the reverb time is close to four seconds or more, instead of just attempting to use high Q arrays, located close to the audience (which still may be needed after treatment), the space should be studied by an acoustician and acoustically treating the room should be considered. After acoustical treatment, the signal/noise (intelligibility), the uniformity of coverage, and sound quality will be superior to results limited only to high Q clusters.

A HANDFUL OF ARENAS

Many sound contractors are aware of acoustical design for recording studios and concert halls, but avoid this work because of its critical nature. Recording studio acoustics are hypercritical. Competition here is keen and the customers have "golden ears." Achieving high definition sound at intense sound levels in tiny control rooms is problematic. Acousticians who conceive these designs, or attempt remedial measures are artists walking a thin tightrope.

Some criteria and guidelines have been established using LEDE techniques which are taught through Syn-Aud-Con and results are becoming more predictable. LEDE tries to standardize (and neutralize) the control room acoustics through control of initial time delay, acoustical absorption of certain directional cues of the room, and manipulation of the reflected sound field's density, temporal elements and intensity.

At first glance, concert hall acoustics may seem less critical, but actually also are very complex problems. Many superior halls exist, and the form factor can be emulated. The rub is that while it is possible to design an excellent hall for an audience of up to one or two thousand, economies of scale seduce the sponsors of new facilities to demand larger and larger halls. Good sounding halls for three thousand or more are very hard to create, and the acoustic compromises of having unamplified music be audible through these facilities are serious. Acoustical innovations used on these jobs have a mixed record and every year one hears of how these halls struggle on with remedial measures. The use of reflective "clouds" have had marginal success and have been shown to have a deleterious effect on low-frequency response.

Cold-Blooded Power

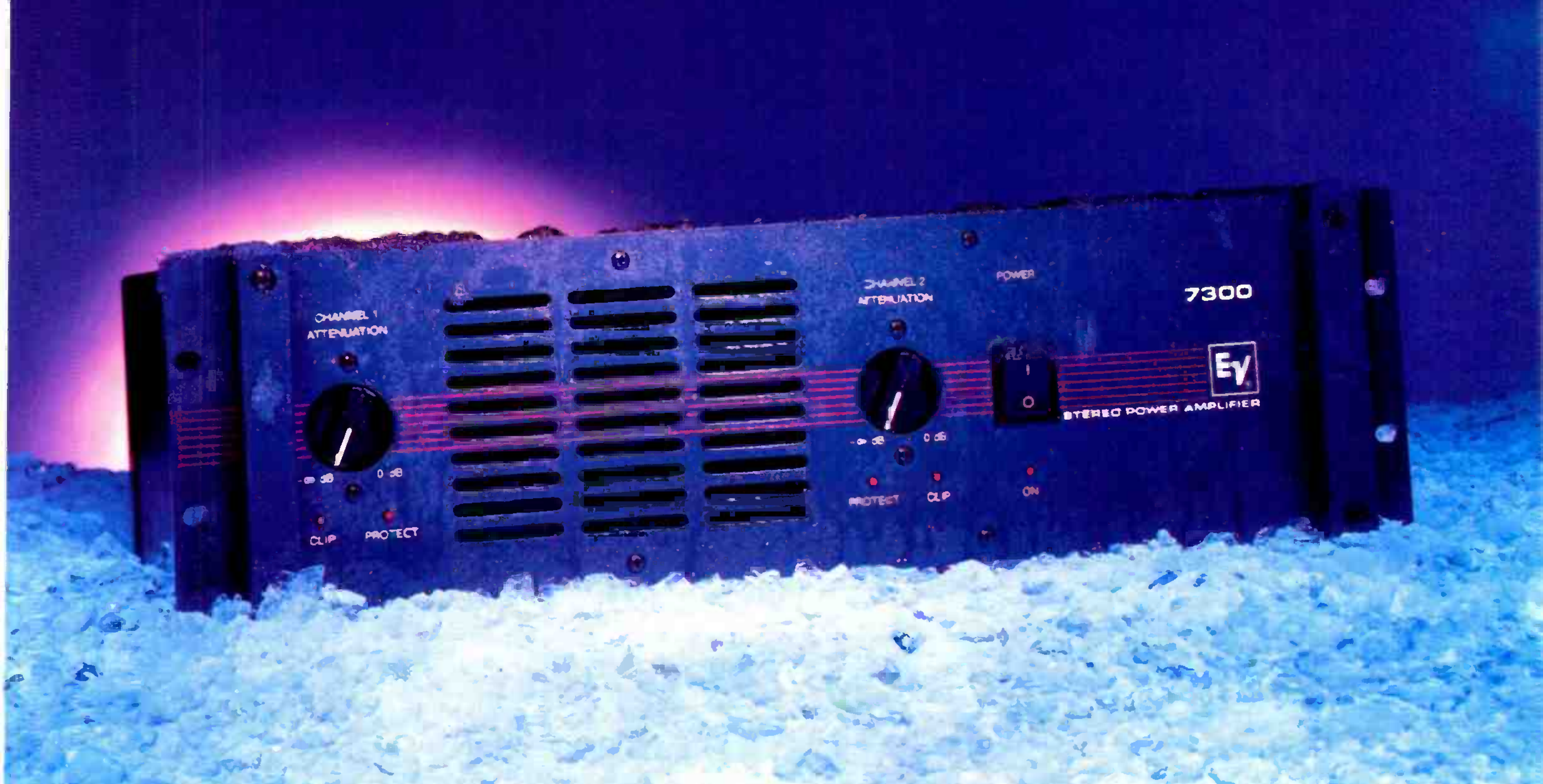

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Figure 1. Wedge contoured 3-inch acoustical foam on the ceiling.

Manfred Schroeder, who headed Bell Labs (and gave us the concepts for electronic reverberators, electronic time delays, pitch shifters, etc.) conceived of a predictable wide-band diffuser of controlled characteristics. Dr. Peter D'Antonio developed and commercialized a family of diffusers based on Schroeder's work. Aside from recording studios, these devices have been used with excellent results in facilities where architectural acoustics, rather than a sound system, is the preferred method of reinforcing speech and music. These wide-band diffusers are very powerful in channeling otherwise wasted or deleterious sound energy into effective use.

Yoichi Ando's research (Concert Hall Acoustics, Springer Verlang) has suggested that non-lateral (ceiling) reflections detract from the aesthetic sound quality. In concert halls, diffusers can effectively re-direct ceiling reflections away from the audience and toward the side walls where this energy will both improve the reverberation density and direction of arrival at the audiences' ears from a spatially more pleasing direction. One of the parameters Ando uses as a figure of merit is the IACC (Inter Aural Correlation), which relates to the correlation of the reflections and their arrival times at each ear. We can see the benefit of re-directing the reflected sound energy from the ceiling toward the walls (instead of absorbing it) helps preserve the limited energy acoustically generated by an unamplified performance.

Figure 2. Wide-band diaphragmic absorbers in the gym.

Ando uses impulse testing and pseudo-random signals as the exciting signal, although it would be possible to transfer some of his work to TEF techniques. Ando believes that by analysis of energy/time/frequency information, and their correlation (using a dummy head), the subjective qualities of an acoustic space can be determined. Ando's work can be used by acousticians as a guide for modeling halls, just as studio designers are using LEDE control room design. While Ando's work is a powerful tool in analyzing energy/time curves, we must remember that a symphony orchestra rarely generates more than an acoustic watt, while a large arena is an entirely different animal.

In a sports arena the house sound system typically has 10,000 electrical watts (100+ acoustical watts) and a rental touring sound system may have 50,000 to 200,000 electrical watts. (It is not good aural hygiene to even think about how many acoustical watts this is!)

Instead of 2000 seats, we may have 20,000 seats. Most arenas were originally designed for sports events such as basketball, ice hockey, or soccer. Architects for these facilities pay little attention to acoustics with the focus on ease of maintenance and durability. Interiors tend to be hard, as acoustically reflective surfaces such as concrete, cement, plaster, steel masonry and hardwood are the dominant construction materials. Seating is often molded plastic. Even the newest facilities that use "acoustic treatment" usually have improved characteristics only above 1000 Hz, with the low frequencies still out of control. Flutter echoes, poor intelligibility, and high ambient noise are the norm. When Ando's criteria are applied to studying the sound field in these spaces, the results indicate why the sound quality is so poor. Crowd sounds are accentuated by noisy, cavernous acoustics. Announcements and narrations of events tend to be fatiguing and musical performances are aesthetically marred and generally unacceptable.


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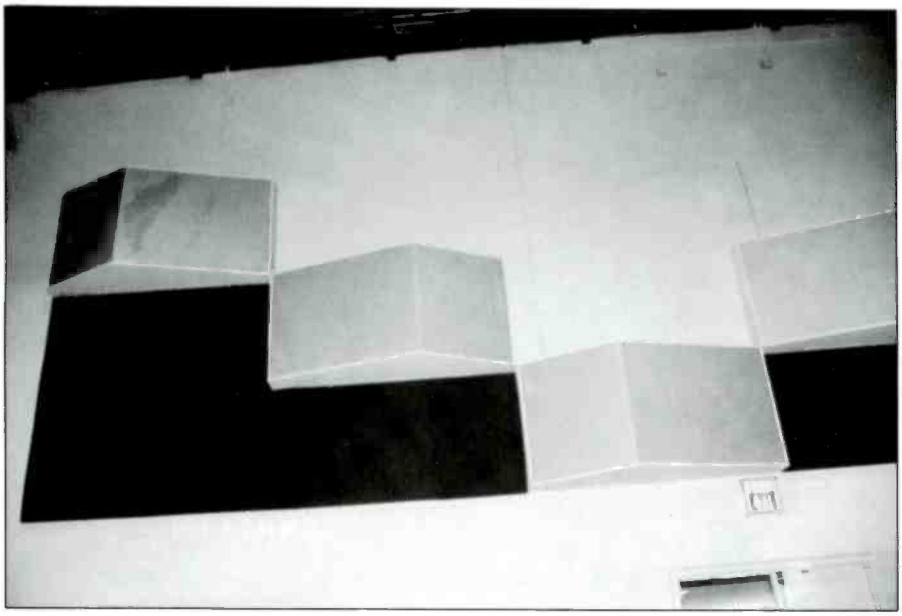

Figure 3. Acoustical foam and diaphragmic absorbers.

While most arenas are designed for sports events, much of these facilities' revenue is now derived from rock concerts. These venues must meet the demands of today's more discerning audiences. The great strides made by the music and audio industries over the last few years had profoundly altered the expectations of the typical concert goer. As a concert ticket costs more than a digital disc, the live concert sound has at least the realism, dynamics and clarity of the disc.

Most competent sound contractors often try to make the best out of a space with poor acoustics. To minimize the room's deleterious effect, highly directional speakers are located as close as possible to the audience. While this approach will improve the ratio of direct to reflected sound reaching the listeners, it cannot help with controlling crowd noise. Poor signal-to-noise ratios will degrade intelligibility and require higher system gain and reduce feedback margins. When the facility is used for concert events, the house system is normally bypassed and the typical rental sound system will result in the usual acoustic chaos. Some sound contractors and touring sound companies have had enough bad experiences with poor acoustics; they avoid these jobs altogether. The comprehensive solution is to control the acoustic space, which will allow more acceptable performance from the wide range of sound system configurations that will be used.

Figure 4. The Digisonix dx-47 Digital Sound Controller.

WIDE-RANGE ABSORPTION

Unlike concert halls, arenas need all the wide-band absorption they can get. Rock concert touring sound systems can radiate hundreds of acoustic watts. Non-lateral ceiling reflections should be absorbed rather than diffused. This is not just to improve aesthetic quality (as in concert halls), but is required to achieve an acceptable signal-to-noise ratio from the sound system over the crowd noise. A sports arena or gymnasium that has been acoustically designed to an optimum specification may have a 10 dB lower reverberant sound field than the typical facility. The commonly used Puertz assumptions for intelligibility criteria use a 25 dB signal-to-noise ratio. Often an untreated facility will have an ambient crowd noise of 80 to 90 dBC, with peak levels of 100+ dBC. This would require the sound system to operate at 105-115 dB with peaks of 130 dB. Although announcers avoid talking over peak crowd levels, sometimes communication under these conditions are critical ("get off the field!").

As a practical matter, sound systems cannot operate at such excessive levels as it would be offensive (at least for ice shows, circuses, sports events and the like). Rock concerts do operate at these levels and here the problem is that the room is not "fast" enough. The decay time of the room must be quick enough that the previous sound (chord, sung phoneme) must have decayed in level sufficiently that the following note can be perceived.

This is discussed in some detail by Fritz Winckel in his book, *Musical, Sound, Sensation* (Dover, 1967). Polyphonic music of the 1500s reflected the cavernous acoustics of the churches of the time, chamber music was obviously named after the intended acoustical environment, and symphonic music intended to be performed in the concert halls of the 1700s. Events move quicker these days and arena architects have not gotten the message yet. When the reverberation of the music and the crowd noise is not being absorbed fast enough by the room, then the music becomes a raucous din. Many theaters of good acoustical design (built in the 1930s to 1940s) are now venues for popular music. Although fan-shaped walls, stepped ceilings, and other acoustical techniques were integrated

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Noel Lee, Head Monster
Monster-Cable Products, Inc.

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into the design by the original architect, the high-power sound systems used today were never envisioned. Just the energy generated from the stage monitor system is beyond the effective absorption of the room (let alone the stage area). Slap-back, unintelligible verses, rumbling, boomy bass and hard, noisy and fatiguing mid-range and highs are the result.

In the case of large facilities that use high-power sound systems, far more energy is typically generated than the space can gracefully absorb. For this type of space, wide-band absorption rather than diffusion is required.

Wide-band absorbers have been developed for control of reverberation and crowd noise in large facilities that use high-power sound systems. Typically installed by sound contractors, they enhance the acoustics of arenas, night clubs, discotheques and gymnasiums.

Existing conventional acoustical surface treatments originally conceived for open plan offices and the like can be supplemented by these wide-band acoustic absorbers. Conventional surface treatments are helpful in increasing the average mid-band absorption coefficient cost-effectively.

THE WIDE-BAND DEVICES

The Cylinder Absorber

An acoustical device specifically engineered for improving the sonic characteristics of large spaces. A hybrid design combining 3.5-inch thick wedge contoured acoustic foam on its circumference for absorption above 400 Hz and Helmholtz resonator tuning for low-frequency absorption. The Cylinder-Absorber achieves 20 Sabines of wide-band absorption at lower cost per Sabine than devices intended for recording studios. Other special considerations for large public space applications are the fail-safe two point Aeroquip rigging hardware and fire-safe Class I/ASTM-E84 Melamine resin wedge contoured foam. Strong, but lightweight construction results in a 45-pound device that will not be a significant structural loading factor.

Most wide-band absorbers use passive low-frequency absorption techniques, essentially relying on the friction or shearing action of dampening materials to convert sound energy into heat. This approach is adequate and appropriate when limited amounts of absorption are required, but not when the output of a 100,000 watt sound system is thundering out of control in a theater, disco or arena!

An alternative approach for low-frequency absorption is the Helmholtz resonator. The Helmholtz resonator has a natural frequency of resonance, and sound is absorbed at this frequency and at adjacent frequencies. A few industrial noise absorbers have been marketed that use this technique for low-frequency noise cancellation, but do not have a wide enough bandwidth (to work with the acoustical foam for wide-band absorption) and have poor transient response. High-efficiency, low-frequency absorption is achieved in the Cylinder-Absorber by slots cut into the six-foot high, two-foot diameter tube. The slots create the Helmholtz resonator effect at the surface of the tube. Glass fiber on the inside and acoustic foam on the exterior provide controlled resistive aperiodic dampening of the resonator Q, so almost an octave of absorption is achieved. The center frequency of the slots is staggered between 30 Hz to 300 Hz. Energy not absorbed by the slots is diffused and energy not absorbed by the wedge-contoured foam is diffused by the tube, or absorbed by the fiberglass within the tube.

Progressively above a few hundred hertz, the absorption is provided by the wedge-contoured foam. To satisfy the rigorous Class I, ASTM-E84 fire code, specifications typically required in large public spaces, Illbruck Illsonic melamine open-cell foam is used (the same material used in Sonix 1). Flame spread of the foam is rated at 5, smoke spread at 65. The foam is cut to a 3.5-inch depth wedge contour which results in an absorption coefficient better than 0.9 from 400 Hz to beyond 8000 Hz.

The Cylinder-Absorber is available in the natural white foam exterior or in custom-painted colors. For applications where the devices will be accessible to contact, a protective acoustically transparent shell of perforated aluminum is available. Guiford fabric exteriors are optionally available for the protective shell version.

Future Developments

Menlo Scientific has been exploring with Digisonix, a division of Nelson Industries, Inc., future potential uses of their digital sound cancellation controller technology in the areas of crowd noise, flutter echos, and reverberant fields. By using this active noise control concept, which is extremely effective at reducing low-frequency noise in ducts, it is envisioned to develop techniques to improve the low-frequency performances in the areas of speech intelligibility. Active cancellation controllers are available for research purposes. As this work continues, progress reports will be published.

While neither Helmholtz resonators nor diaphragmic absorbers are news, both these devices are the first commercially available, "industrial size" wide-band devices built specifically to be cost-effective for large facilities that use high-power sound systems. The next device, the digital signal processor noise canceller, is brand new.

The unique aspects of acoustical design for large facilities with high-intensity sound systems have been discussed. Solutions have been successfully implemented in a number of facilities using wide-band acoustic devices in conjunction with surface treatments. Future developments using electronic cancellation of crowd noise and/or far-field reverberation has been introduced.

ADDITIONAL READING

Donald Davis, "Design Room or Sound System?" Project Notes/Engineering Briefs, April 1973.

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Yoichi Ando, "Concert Hall Acoustics," Springer-Verlag, New York, 1985.

Leo Beranek, "Concert-Hall Acoustics: Meeting Place of Science and Art," Physics Today, December 1986.

Fritz Wrinckel, "Music, Sound, and Sensation," Dover 1967.

Michael Klasco and Rudy Paolini, "San Diego Sports Arena," db Magazine, November/December 1987.

David Scheirman, "Reinforced Audio—Better Sound for Arenas," Sound & Video Contractor, January 15, 1988.

THE IEQ WITH Smartcurve™

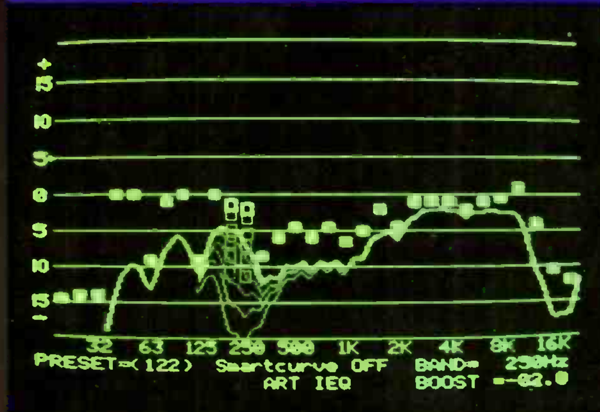
Not Just Another Programmable Equalizer With MIDI™

The IEQ with Smartcurve™ is a programmable, high performance graphic equalizer that includes a video output. For those who wish to enjoy the video output of the IEQ, ART makes the IEQ Video Monitor.* Smartcurve™, proprietary software developed by ART gives you instant actual frequency response as easy as the push of a button.

The IEQ Family consists of both the 2/3 octave and the 1/3 octave graphic equalizers. Both types come in two varieties, Controllers and Satellites. A Controller is a self-contained programmable intelligent graphic equalizer capable of controlling 15 satellites at once. IEQ Satellites are exactly the same unit except the front panel controls are eliminated.

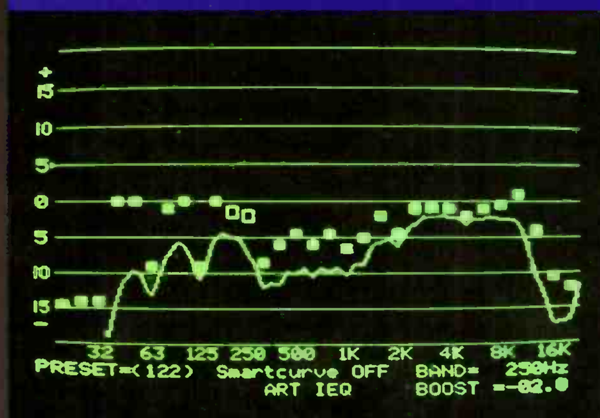
IEQ Model Specifications:
Controller & Satellite
- 128 battery backed presets
- MIDI
- Frequency Response 20Hz-20kHz ± 0.5dB
- THD · ≤ .009% @ 1kHz, 0dBM typical
- Dynamic Range · ≥ 100dB typical
- Balanced inputs and outputs

***IEQ Video Monitor Features**
- 19" rack mountable
- NTSC compatible monochrome monitor
- 4 Selectable inputs
- Standard RCA jacks for easy connections



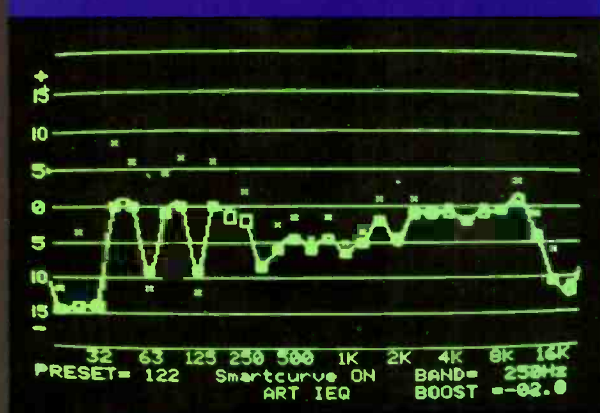
1 See the Sound

This is a video output of the IEQ as the unit is being adjusted. The sliders can be moved ± 15dB in 1/2dB steps to get the exact response you need. With the simple push of a button, complex equalization can be done in seconds with incredible accuracy.



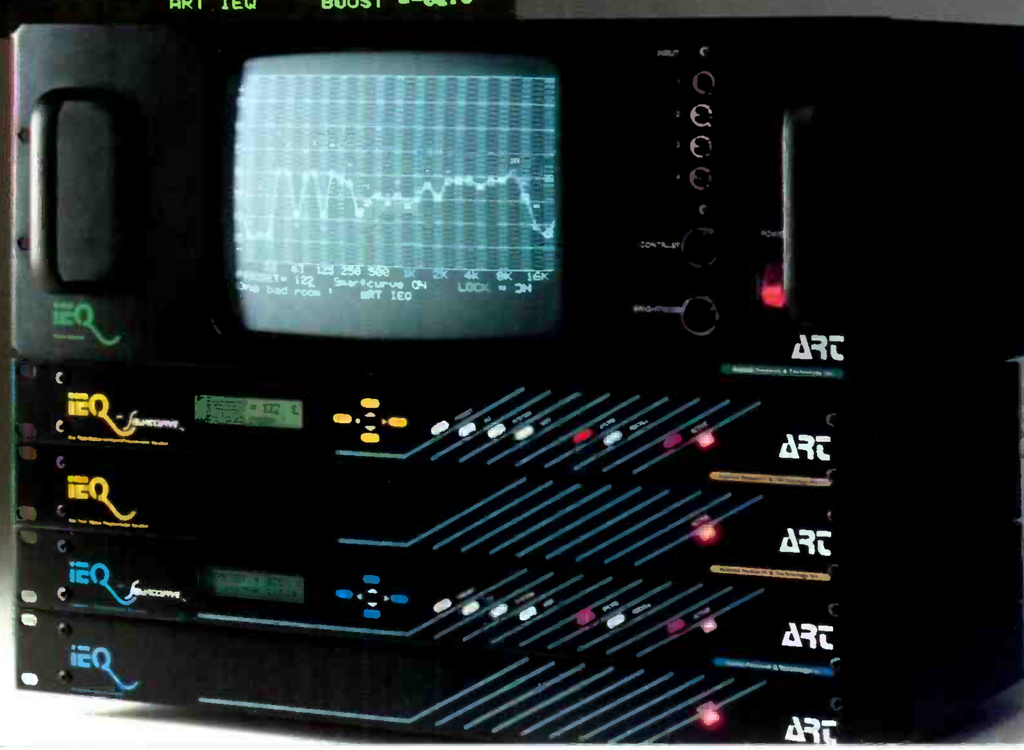
2 Hear the Sound

The power of the IEQ readily becomes apparent as the video display plots the frequency response due to the slider settings. The IEQ offers high quality constant "Q" equalization. The video graphic display shows the correlation between the sliders and the frequency response.



3 Perfect Sound

Turn Smartcurve™ on and perfect equalization is at your fingertips. Note the difference between the second frame, (Smartcurve™ OFF) and this frame. The position of the sliders represent the actual frequency response of the EQ. Interaction between bands is virtually eliminated. Incredible! Just think, now when you adjust the EQ you get exactly what you need. The "perfect" EQ? Let your eyes and ears decide.



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ADDENDUM

Specific problems of ceiling, wall, and seating constructions can be identified and remedied.

Ceilings

Non-lateral reflections are not constructive as most arena ceilings are over 80 feet. These (not-so-early) reflections reduce the clarity and definition of both speech and music. Lobes from the speaker clusters reflect off the ceiling and reflect back into the seating areas. If the ceiling is acoustically absorptive then the first reflection off the ceiling will lose most of its energy before it reflects its way back into the seating. Even if the ceiling is absorptive only in the mid and high frequencies, this will still help shape the reverberation decay contour. Remember, if the mid/high frequency energy is absorbed on the initial reflection, then regardless of the absorption coefficient of subsequent reflections, the mid/highs will not be reflected. Time/frequency/energy spectral decay curves are a useful tool in analyzing the intensity, density, and spectral distribution of the early reflections.

TYPICAL CEILING SURFACES

Ceiling surface materials are often corrugated or perforated metal, or "acoustical" ceiling tiles, although sprayed-on treatments are sometimes used.

Corrugated

Some mid-range diffusion occurs due to the shape, the absorption is negligible. Glass fiber is often located above the corrugated ceiling, but this only aids the thermal properties.

Perforated Metal

Glass fiber is typically located behind perforated panels, but the perforations usually account for less than 10 percent of the surface area, with only marginal absorption benefits.

Acoustical Ceiling Tiles

Conventional tiles have little absorption below 1000 Hz, and generally only fair characteristic in the mid-band. If the material is less than 1-inch thick, it is not going to be highly effective.

Plaster/Concrete

Some high-frequency absorption (especially from unpainted concrete), but otherwise all energy will be reflected.

Spray-on Surfaces

Asbestos can still be found in sound facilities, but are often painted over to seal the material. Asbestos has been found to cause cancer, and is no longer legal in new installations.

House sound systems often have lobes that reflect off the ceiling and back into the seating areas, degrading the clarity and definition. Even if constant directivity horns are used in the house system, sound systems brought in by touring sound companies are unpredictable. Crowd noise will, of course, be totally unaffected by whatever sound system is used, and a highly absorptive ceiling will help improve the signal-to-noise ratio of the room.

CONVENTIONAL ACOUSTICAL TREATMENTS FOR THE CEILING

Non-asbestos Cellulose Fiber spray-ons

These are cost-effective and have good performance but care must be taken that fire code requirements are satisfied. The ceiling must also be an appropriate substrate or chunks of the material may delaminate. Ceilings that flex, such as most corrugated metal, are questionable candidates for spray-on treatments. Good mid-bass performance requires at least 3-inch depth, which means multiple coatings will be required. Proper resin content and careful control of thickness is necessary to avoid serious maintenance problems. In warm and humid climates where ceiling sweating is common fungus and adhesion can be problems. Assuming your installation is appropriate for spray-ons, the only separate low-frequency absorption devices will be needed to provide wide-band treatment. Spray-on materials can also be used for wall treatment, but they must be out of reach as they are not robust to contact.

Glass Fiber

This is the commonly-known treatment, as it is relatively inexpensive, meets fire codes, and does not degrade over time. The hidden cost is that it is rarely usable without some additional material fabrication such as perforated metal (60 percent open), or fire-retardant fabric covering. One trick to keeping cost down is to use panels of semi-ridged, black-dyed glass fiber, attached to the ceiling by bolts, with large (3-inch) diameter washer to catch the material. If the ceiling is high enough, the appearance may be acceptable, the performance is good, and the price is right.

Like other surface treatments, absorption for 3-inch semi-rigid glass fiber falls off below a few hundred Hz. Careful design of spacing the panels slightly away from the ceiling can improve the low-frequency characteristics through diaphragmic absorption.

Acoustic Foam

Various suppliers of acoustic foam market convoluted or wedge-shaped panels. For really effective absorption into the voice range panel thickness should be 3-inch or greater.

Most foams do not like ultra-violet or moisture. Non-bridging paints that offer improved fire retarding and better smoke spread ratings can be applied. Halcon coatings are available to eliminate moisture problems, but some high-frequency absorption is lost.

The big problem with foams are fire codes. Installations that require over five-thousand square feet of treatments or have a ceiling less than 25 feet will rarely be acceptable to the fire marshal.

Melamine Resin Foams

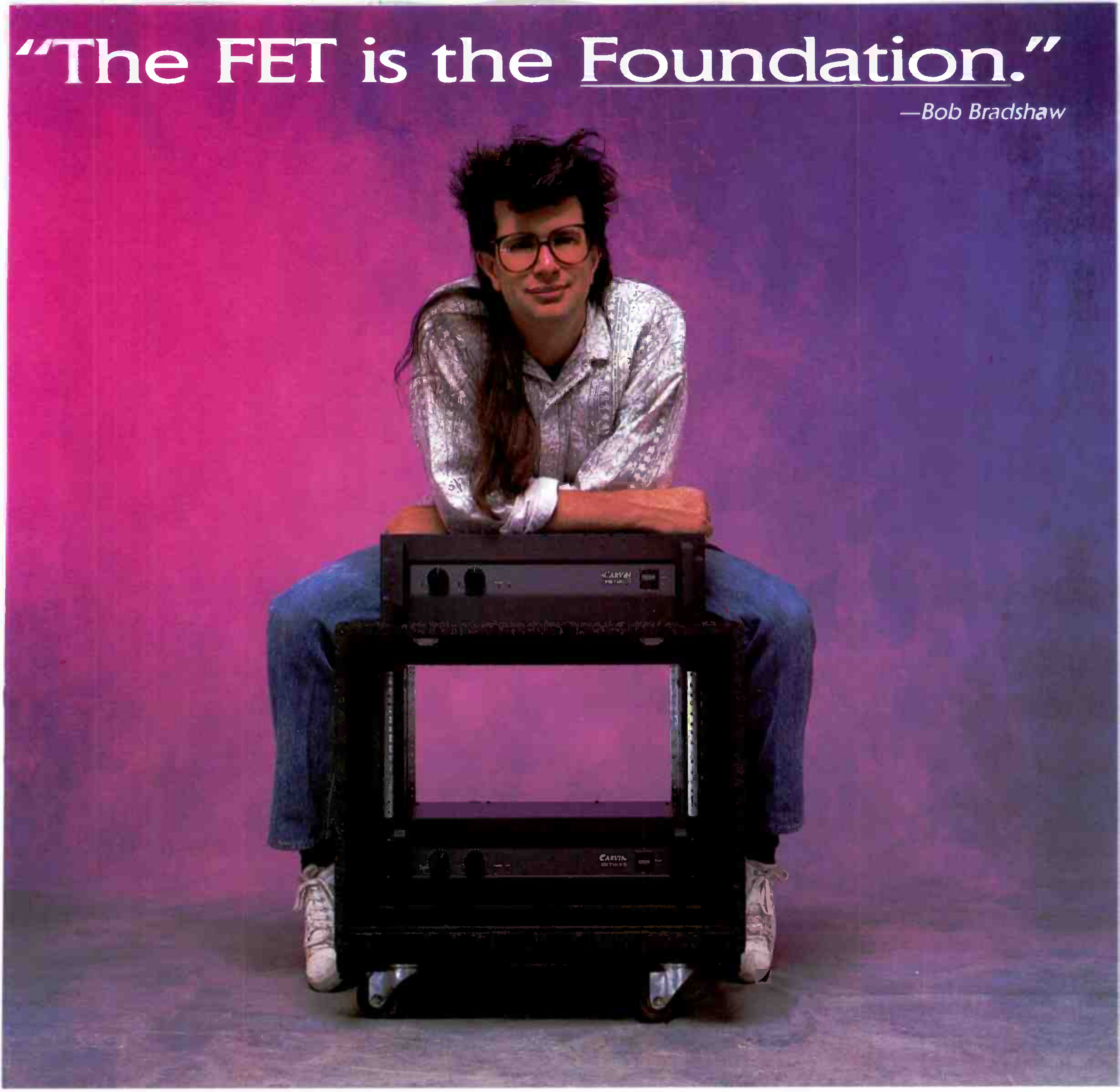
These materials offer superior absorption characteristics (especially between 250-500 Hz) compared to conventional foams, meet the rigorous class I and ASTM-E84 fire-code specifications. The price is almost ten times higher than raw glass fiber.

WALL PROBLEMS

Parallel Walls: Large expanses of hard-surfaced parallel walls that are not angled or splayed will create flutter echoes.

Curved Walls: Will refocus the sound creating uneven coverage and slap-back.

Most of the treatments discussed for ceilings apply to walls, but none of these surface materials are effective below a few hundred Hz nor can these materials be used within reach



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WIDE-BAND HIGH ENERGY ABSORPTION DEVICES

Menlo Scientific Diaphragmic Absorber

These devices are wall-mounted and are five-sided, with an off-center pointed front section. Perforated steel screens are used (with optional fabric covering).

Internally, a diaphragmic membrane is used to absorb low frequencies, and two-spaced layers of semi-rigid glass fiber absorb mid-bass through the high frequencies. The glass fiber is spaced to aid low-frequency absorption. The steel screens work to diffuse whatever energy is not absorbed. The construction uses specially treated fire resistant wood.

RPG Abffusor

An absorber-diffusor for medium-sized facilities, high absorption is achieved down to 100 Hz when installed with a 12-inch air space behind the device. The device consists of an array of wells of equal width, but different depths, separated by thin dividers. The well surfaces and dividers are fabricated from a porous absorptive material.

Menlo Scientific Cylindrical- Absorbers

Cylindrically-shaped Helmholtz resonators 2.5 feet in diameter and six feet long are used to provide low-frequency absorption. Slots cut into the 6-foot tube provide low-frequency absorption. Glass fiber on the inside of the wedge and 3-inch thick wedge-shaped acoustic foam on the outside provide aperiodic dampening of the resonator Q so almost an octave of absorption is achieved. The tuning of the cylinders is staggered so when multiple units are used, absorption is effective between 20 and 200 Hz. Above this range the absorption is crossed over to the acoustic foam on the surface of the tube. The foam is a special melimane resin that meets the ASTM-E84 Class I fire code standards. The foam is cut to a 3.5-inch deep wedge contour. The cylindrical absorber has been engineered to diffuse whatever energy that it has not absorbed, although its absorption coefficient is close to 1 over the mid and high frequencies. db

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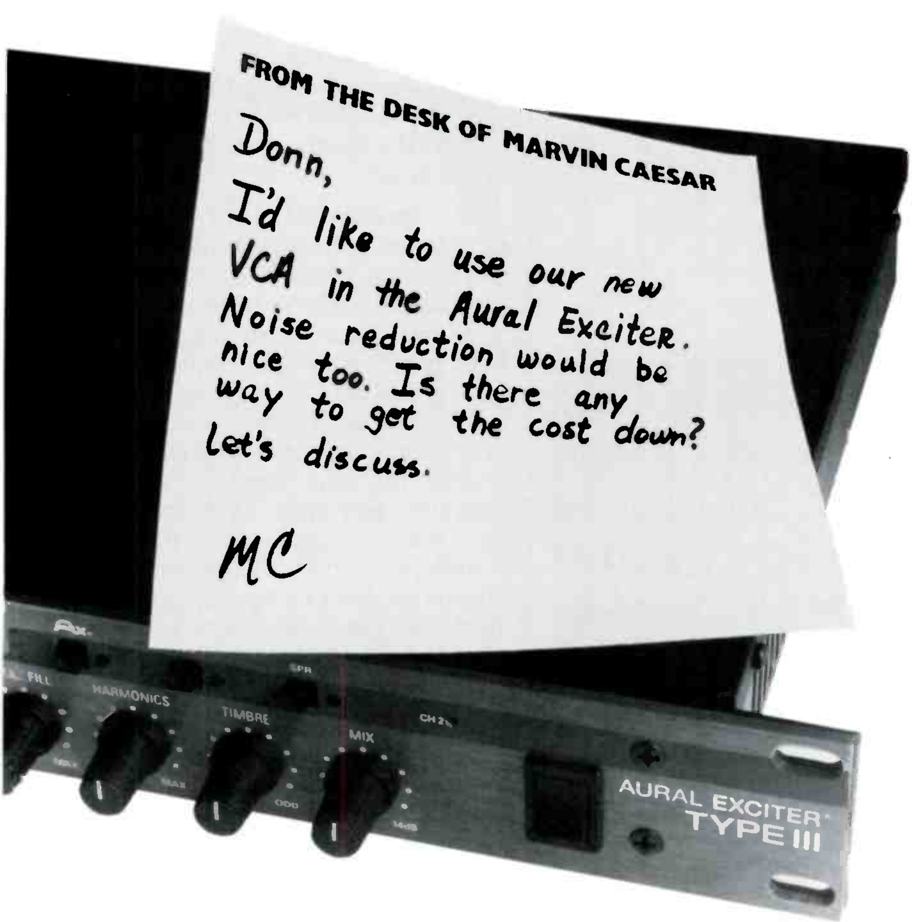

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On Tour with the Dixie Dregs

The subject of this story is *sound reinforcement*. Reinforcement simply means, according to Webster's: "...the action of strengthening, making stronger or more pronounced."

Notice how there is nothing stated in this definition that implies an alteration or change in the structure of the *reinforcee*. One definition that I like is: To make bigger than life. Bigger is probably the best way to describe it.

The problem is that the reinforcement of music is not easy. The sound engineer is at odds with many variables such as the size of a room and its acoustics (reflections, dampening, resonances, traps, decay times, directivity, dispersion, nodes and diffractions, not to mention the ominous *eighth space*) that will all have a direct impact on the quality and intelligibility of the music.

However, this story is not about acoustics. It's about the music and the way that it comes across to the listener.

Corey Davidson is Technical Editor of db Magazine.

Sure...it *is* very important to understand acoustics, but an understanding of music is what lies underneath all of the technical jargon.

When you combine the *dynamic range* of the New York Philharmonic with the electrical power of Van Halen, then fold in the musical styles of jazz, country, classical and rock, you will most likely come up with the music of **The Dregs** (alias **The Dixie Dregs**). Many believe that the music of The Dregs is some of the most challenging electric music available today—what a good basis upon which to establish a story about sound reinforcement.

THE DREGS

Our interview is with sound engineer/system designer Ken Hirsch. Ken tells me, "This band (Dregs) is one of the most demanding groups of artists that I have ever worked for. These musicians are the pinnacle of their respective categories in music. They have all won awards (and Grammy nominations) for their instrumental prowess, both individually and as a band."

Figure 1. Ken Hirsch with Shubert Systems Group equipment.

I certainly shared those perceptions for I had worked as synthesizer programmer for T. Lavitz (Dregs' keyboard player) both before and after his membership in the band. My job in those days was to recreate sounds that T. was interested in having on his synthesizers. He would bring me tapes of other bands and ask me to get the sounds that Keith Emerson, Herbie Hancock, Jan Hammer or Chick Corea were getting on their synths. Sometimes T. and I would sit for hours tweaking in sounds that he envisioned in his mind's ear. When he changed over to another type of synthesizer, I had to aurally duplicate those sounds on the new synthesizer. Then and now, T. Lavitz is one of the more fastidious talents today. Since those earlier days, *interfacing* and *RAM* have made programming much easier and T. Lavitz has come into his own as a renowned synthesist. Together with his fellow Dregs, they have brought to the world their own special brand of classically-countrified, fusionesque, instrumental rock and roll.

On a Saturday in July of this year, I went to see my friends play one of their last shows as a band. Steve Morse had just finished a stint as a commercial jet pilot and T. Lavitz had contacted the band members for a final reunion. My early arrival enabled me to sit through a sound check...a situation that many find boring. I had the opportunity of hearing the band many times in the past, but I now found myself mesmerized by a drum sound-check. Rod's kit sounded so tight and I know that he is a stickler for great kit sounds. However, the sound out of the P.A. was record-like. Now I had heard many shows at this particular club (including having played there myself) and the house system had not been modified in any way for this show. I thought that the sound engineer must have been using triggered samples of great drum sounds as well as the sound of the kit. I was certainly mistaken. This incredible drum

Figure2. The house system at the Trancus night club.

sound was just *the drums*. I now knew that I had to have a chat with their sound engineer. Anyone who can get such great sounds must know something worth sharing.

PERSONALLY MODIFIED GATE

I met with Ken on Friday, August 19, another one of the Dixie Dregs' last New York appearances. It was before their final show at a venue called Sundance, in Bayshore, Long Island that Ken expressed his deepest thoughts on sound reinforcement and shared one of his design-modifications of an audio gate.

Figure 3. The 24/16 remote recording van owned by Dennis Dragon.

What kind of modification did you perform on what gate? "An old Omnicraft GT-4 was the gate that I modified. About four or five years ago this gate was available for around $249.00...one of the lowest-cost gates on the market. I had gotten particularly frustrated with them because they were only an on/off gate. I needed a way to control the amount of gain-reduction. I decided that I should simply try to sell the gate and purchase the unit with which I would be happier. I became even more frustrated when I tried to sell it because nobody wanted this gate. Even when I got a bite and somebody would come and see it, they would reject the unit for either not having the features of gating that they wanted, or that the connectors on the back were undesirable, or both. I now had to decide whether or not to trash the unit or get my dollar's worth by modifying it...providing that the modification was cheap."

"The gate had THRESHOLD and RELEASE time. What I wanted to do was to control the amount of gain reduction so that with one knob, I could set the unit wide open for setting up the gates. This way the THRESHOLD was always set and I could open up the gate to get it out of the way. Then I could take the new gain reduction control and turn it down to various amounts of gain reduction so I could drop down 10 dB for just *ducking*. With my modification, the gate became a ducker in addition to being a straightforward noise gate."

Let's describe to our readers a little bit about the principles and operations of a gate such as: the advantages of using gates, and how they are applied.

"Ninety-nine percent of the time I will use gates on tom-toms. To get tom-toms *right* in the mix, they have to be especially prominent, because when that drummer hits them, you want them to jump out. Generally speaking, when a drummer goes for tom-toms, he is trying to make a point, so as a sound engineer, you want them to be right up-front in the mix. Tom-toms are a drummer's way of getting a few licks in much the same way that a guitar player gets his licks in. They often occur between vocal parts. In between vocal passages, a drummer will throw a few licks in just to accentuate his parts rather than playing the straight beat. So I generally bring the toms up. However when you bring the toms up in volume without gates, all the extraneous noise that is on stage or near the drums will be amplified as well as the sound that you want in the mics. This extraneous noise will mess up the snare's EQ, the cymbals, and will add unwanted sounds."

Does the fact that you use gates mostly on tom-toms mean that you try to avoid the use of gates in other areas? "If I've got it available, I like to use gates on kick-drums. Occasionally it really helps to tighten things up. I certainly shy away from using a gate on a snare drum. Most drummers that I work with are sophisticated enough to play paradiddles and second-line beats, and all sorts of wild stuff that a gate would chop up. The basis of a live rock band comes from where the drummer is going. In other words, the rest of the band is responsible for shaping the music around the rhythm. Now this statement might get me into hot water with lead guitar players, keyboard players, and bass players, but let's face it...drums are a priority. A poor drum sound can destroy the overall sound of a band. There is so much enharmonic information in drums that they can easily become too cumbersome and overbearing. If anyone doubts that, all they have to do is listen to any record in popular music. It will be quite clear that a good drum sound can help to define the direction of the music."

Tell us about this tour. "Everywhere we have played (worked) I have had to deal with the house systems. We are not carrying our own P.A. Actually I'm not doing as bad with this tour as with the **Blues Busters.**"

Who's in the Blues Busters? "T. Lavitz (from The Dregs), Larry Zack (from Jackson Brown and Rare Earth), Freebo (from Bonnie Raitt's band), Catfish Hodge (singer), and Paul Barrere (from Little Feat). The current line-up has replaced T. Lavitz and Paul Barrere with Brian Auger (Brian Auger's Express) on keyboards, and Leo Nocentelli (from The Meters). The venues that were booked for The Blues Busters were lousy clubs with terrible P.A. systems. Basically, that was a band of great musicians that had to play in crummy rooms. The tour was certainly a challenge because I had to represent the band to the audience...and the audience is not interested in excuses as to why the sound system might be deficient. Those kinds of gigs are the ultimate challenge. Nothing is more satisfying than to squeeze the most that you can out of minimal gear. Sometimes, having to do that will bring you back to the basics that one must never lose sight of."

"This tour with The Dregs is being booked in venues that have adequate systems, however, it's not easy to use a different system every show. The only advantage is that the P.A. is already in place...a luxury that I gladly would forfeit in order to use my own system. But again, on this tour, we had no choice."

USING HOUSE SYSTEMS

Tell us about your own system. "My system is presently in a night club called Trancas in Malibu, California. That system consists of a Peavey 1608 monitor console, MXR EQs, and Peavey power amps. There are other assorted brands, none of which are ultra high-end brands. The funny thing about that system is the fact that it's piecemeal. Most of what I have is quite affordable to the average person. Generally, my gear is all used and has come second-hand from bands and sound companies."

Do you have brand-name choices? "Not really. I mean, I certainly know what the high-end gear is, but I know that for live sound, headroom is by far, the most important factor. As a matter of fact, it is how well one understands headroom that enables the engineer to get the most out of a system. Beefy transformers (heavy and well-wound) and good power supplies are at the heart of good electronics. With a little investigation, one can learn about what is in a power supply and immediately begin to apply that knowledge to the understanding, purchase, and use of equipment."

What is the average size room that we're dealing with in this discussion? "Generally 500, 600 seat venues."

GAIN STRUCTURES

How much power do you hope to run in rooms that size? "For those sized rooms I would hope to run at least 5,000 to 7,000 watts or more."

How much of that available power do you actually use? "Depending on the gain structure, 9 times out of 10, I'm using almost all of the amp's available power. But I never attempt to clip the amps. The key is to have enough headroom so that you never have to heat those amps up to the point where they are struggling or it will show in the music. You want enough headroom so that it can be as loud as you can stand it. If the band is good and you have done your homework in terms of room EQ and gain structures, it can be pretty loud and pleasurable."

Could you elaborate on what you mean by *gain structures*? "One must learn from experience how preamps react, and which mixing boards can tolerate a clipping indicator that is *on* at a steady beat and which ones won't. Another key is to get the subgroups and the output masters so that you're not driving one or the other too hard. Some boards will indicate a clip to the input of a bus. This is possible as opposed to the indication of output bus clipping. As you pile on gain from the individual channel faders, you can clip the input to the bus. One must be very careful and learn to understand the way in which individual pieces of equipment achieve gain."

"Once you have looked over the block diagrams of two or three mid-range 24x8 to 32x8 channel consoles in the 10,000 to 20,000 dollar range, they start to look very similar. It becomes a matter of your skill in recognizing each console's limitations and its characteristics of overload. There are some economy boards on the market that are very touchy and will go into clip when a very fine line is crossed. Others can seemingly tolerate hot signals that the board sees as 'in the red.' A good example of these two conditions are the old Yamaha PM1000 and a Yamaha 2408. The PM1000 was considered to be, at one time, the absolute workhorse of the industry. With transformer *ins* and transformer *outs*, its broad dynamic range, LC type EQ, and super high output, the PM1000 was a difficult board to clip. Signals could be placed in the red without the slightest aural indication that clipping occurred. This was partly due to the elaborate input pad design. The 2408, on the other hand, is very sensitive to improper gain structures and one will find that suddenly, unless they know that particular console, clipping and distortion are easily encountered. I'm not saying that one is better than the other. On one hand the PM1000 might be desirable. On the other hand, the 2408 is far more versatile in terms of patch-points etc. The point of all these comparisons is to give the engineer the ability to understand the situation at hand. We're talking about making the most with what you've got."

Figure 4. The challenge presented by a large band of horns. An array of horn players mixed by Ken Hirsch. The photo includes players from Tower of Power, Mad Dogs & Englishmen, Rod Stewart, Ricki Lee Jones, Disney sound track orchestras, and Barry Manilow.

"A good rule of thumb is to look at the power ratings of a given piece of equipment and realize that the manufacturer generally under-rates. This can be to a sound engineer's advantage in order to find new levels and limits. Many speakers can take two and three times their rated power limits. I have never blown a speaker from too much clean power. You can. But what I'm saying is that distortion from clipped signals will do far more damage, even at lower levels, than just too much power. Clipping will cause things to heat up sooner. Clipping at high power levels is disaster...both for the equipment and the listeners!"

When it comes to reinforcement, there are so many factors that will affect the delivery of sound. A difference between 16 kHz bandwidth and 20 kHz bandwidth will not make or break a show. Ken adds, "If I were to compare a Peavey stack, properly set up and properly powered, to a JBL stack, properly set up and properly powered, the differences are minimal."

"The real test is to be thrown into a situation with nothing but your hands and the gear that they throw at you, and still be able to pull-off something that really sounds good. That is the mark of a truly good sound engineer. It's impressive when someone can walk up to an entire system that they have never seen or operated before and make it sound as good as someone else who owns the gear."

INSTRUMENTAL VS. VOCAL

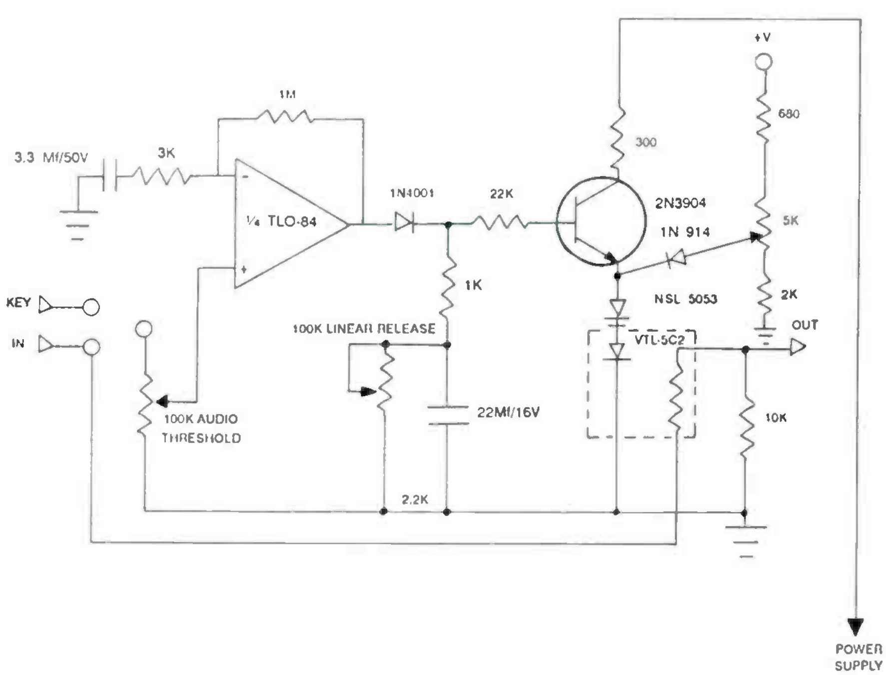
What's special about sound for The Dregs and is there an advantage to the fact that they are not a vocal band? "Equalization for a vocal band as opposed to equalization for a purely instrumental band are two completely different animals. In reinforcing a vocal band, one must leave a humongous amount of gain and bandwidth for the vocals. You must clear away a lot of the frequencies of the instruments that would override or interfere with vocals. With instrumental music you've got the entire bandwidth to fill up with all the various instruments."

Do you prefer one over the other? "I like both. When I first started out doing music, I worked with primarily country and country rock bands. I cut my teeth working with bands that had four and five vocalists and learned to run the entire band with one hand, and all the vocalists with the other. I actually had each finger on a vocalist. Instrumental bands are refreshing in that you can concentrate on instruments and let *them* be the stars."

How is recorded music and its live counterpart related? "Live music is essentially an exaggeration of what you hear in recorded music. The vocals will be louder. The drums will have more dynamic range and will jump out. The leads will be louder. You're really exaggerating the truth that the record started out with...at least that's the way my philosophy works in mixing. The bottom line is that you're cutting a fine line between truth and a characterization in order to bring out the highlights and make them clear to a live audience."

"When a guitar player walks up to the front of the stage and starts wailing, the audience expects it to be bigger than life. Typically, on a record, the guitar (at this same point in the music) would be

Figure 5. The final version of the Omnicraft GT-4 modification.



PA Gear Pr

Powered Mixers

Model PM-1600 gives you everything you need in a state-of-the-art PA mixer — 16 INPUTS, 2 STEREO POWER AMPS, DIGITAL EFFECTS PROCESSOR, TWO GRAPHIC EQUALIZERS **AND** A 2-WAY ELECTRONIC CROSSOVER — in just one performance package! Imagine, no more lugging around separate units, no more wiring hassles and no more impedance mis-matches!

Unlike other PA mixers, this has not one, but *two stereo power amps* built-in. These are not some "second rate" amps, either. The first amp is basically the N.I.H. Model P-700 described below. 400 watts per channel into 4 ohms! The other stereo power amp puts out 250 watts per channel into 4 ohms. Since these are **stereo** power amps, you have 4 channels of built-in amplification you can use *any* way you wish.

The built-in digital effects processor has 99 programs including digital reverb and digital delay. There are LED meter displays for the mains, effects buses and monitor, a built-in talkback mike, a stereo 9-band graphic equalizer for the mains, another 9-band graphic for the monitor and a whole lot of patching flexibility. This is your best bet yet in a PA mixer. **SAVE $2,755*!**

Circle 21 on Reader Service Card

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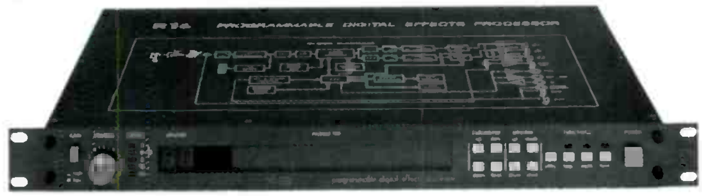
they do in world class studios and sound reinforcement companies. With "Q" or bandwidth controls, you can dial in the exact EQ sound and range you need without affecting tones that are octaves away from your center frequency. With the PE-30, you control the notch. **SAVE $396*!**

Model R-16 is the second best digital effects processor in the world and about 1/10th the price of the very best (the Quantec QRS/XL). There's a 16 bit processor PLUS a 32 bit numeric co-processor for **superior** reverb sound quality; full



MIDI control of effects parameters and changes; sampling capability; 99 programmable sounds (30 pre-set); user changeable parameters. This unit eats all other digital effects processors for breakfast, except for one. **SAVE $1505*!**

Now that you've had a taste of the good stuff from N.I.H. Labs, here's the real clincher:

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P-700	Integrated Power Amp	$1,300.00	**$ 699.00**	$ 601.00
CN-400	Crossover Network	$ 850.00	**$ 349.00**	$ 501.00
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mixed back in so that it blends because the definition of the average home stereo system is considerably better than a live P.A. system. In the home stereo context, you're not working in a hostile environment like arenas, high school gymnasiums, field houses, etc."

How do you feel about real-time analysis of a space, either before or during a performance? "I've tried RTAing rooms. First of all, it annoys everybody involved because you need the room to yourself with this hideously loud-wooshing pink-noise assaulting the ears of anyone within the space. RTAs are effective to a certain point but they don't tell you the impulse response of a room."

Explain, please. "When a snare drum is hit hard, the room responds in a very different way then when a singer sings a note. So there is an impulse and a steady state signal. A steady state signal will eventually fill a space and build up nodes. An impulse sound will smack straight out into the hall and bring back certain resonant frequencies. Generally, pink-noise analysis will show you where the room is building up nodes, but those nodes interact and create new nodes."

"An RTA is a wonderful tool...if you can learn to ignore it when you have to. It's like having an oscilloscope for analyzing waveforms...it's wonderful if you need it. When you're doing microscopic analysis, scopes and complex test equipment can certainly help to explain and correct problems. Those people who have vast experience in electronics know that you can get away with just a volt meter and get 90 percent of what you need to know. The volt meter in this discussion could be analogous to your ears and the meters on the console. Personally, I own a Gold Line 30 real-time analyzer. The mic that I use with it is not the calibrated mic, but it's a mic whose characteristics I know, and it can take getting beaten and stepped on. If I see (hear) anomalies that are boggling my mind, and I find myself groping in the dark for EQ, the RTA can help find an answer."

"To answer your question as to whether or not I use RTAs, I really don't believe that it offers me an advantage. I prefer to start with an album or record that I have played on my own, personal stereo system. If you take a CD and put it in anyone else's CD player, it's going to sound different. CD players vary from one make and model to another because of the differences in A/D, D/A conversions, and filtering. So I reference P.A. systems with what I know I'm familiar with in a previous controlled environment. If I know what it should sound like on my personal system, I can take that same material and go for as good a duplication of playback as possible on a P.A. system."

HANDS-ON SUGGESTIONS

Do you have any suggestions as to how a beginner might become familiar with reinforcement in a hands-on context? "Start working in a local club on a regular basis, or start working for a sound company. Now I know that this may sound like one has to be a grunt and sweep floors, but sometimes we must all pay a price to get to the next level. If you want it, it's always worth a sacrifice. It is very important for the beginning engineer to work with nothing (as little as possible). No gates, no compression. Learn how to ride vocals by hand. Try to familiarize yourself with as many processing devices as you can...although that's becoming impossible...I don't know half of them at this point in time so I try and stay with the ones I know. Try learning how to mix and work without aids or luxury tools. Don't go out and buy a pile of gear and then sit there and spend all of your time learning how to operate it when your priority should be *what the band sounds like.*"

For our techy friends out there, let's look at your gate modification (*Figure 1*). "A reverse-log or audio taper potentiometer is the pot of preference for this gain modification. I used an RV5 pot, which is a very small pot that has the leads coming off the back. If the leads were to have come off the side, it would have interfered with the pot in one direction and with the case in the other direction. It is necessary to generate a voltage divider around this pot. The top 680 ohm resistor drops the voltage down from +V so that the emitter of the transistor is not pushed too hard. The LED in the opto-isolator is also protected by the drop in +V. The 2k resistor biases up from ground enough so that the voltage potential is not dragged down too far. Basically, 2k ohms is where I get no more of the ducking effect...so that is essentially infinity."

"The diode (1N914) prevents an improper voltage feedback. This diode sits between the LED of the front panel and the emitter of the 2N3904. An elegant aspect to this design is that the audio path is traveling through only an L-pad. One part of the L-pad being the opto-resistor and the other part of it is a 10k resistor to ground, which essentially gives you a variable L-pad. When the opto-resistor is totally on, you have zero attenuation. As the LED in the opto-resistor dims out, it increases in resistance and turns into an L-pad of increasing attenuation until infinity is reached. This is how gain reduction is achieved without the use of an active component...like in a VCA, which would drastically increase the cost of the modification."

"The added bonus is that the LED on the front panel follows the amount of gain reduction, so as you turn the gain reduction to increasingly lower levels, the LED on the front panel will light up, indicating that you're ducking a little lower. When the LED is fully lit, the gate is wide open. As you dim it down, it will indicate the reduction. When the gate opens, you'll see the LED get brighter. Again this is a very simple modification that helped save this piece from atrophy. The most difficult thing here is drilling a hole in the front panel and charting your amount of gain reduction, which I did with a standard loss meter that I got from a local telephone company."

In closing, Ken adds a strong voice of inspiration, "One of the things that happens to many people as they become exposed and involved with this part of the industry, is that they discourage themselves by saying: '...it's so difficult and complex!' But it's not that complex. Think about magic. You look at a magician and he does this incredible trick. You're sitting there totally amazed, wondering how it was done. For the few people who have actually learned how the trick was done, they are surprised as to how easily the trick is accomplished. It's always amazing and awe inspiring until you know how it's done. Sometimes you can amaze yourself when you get good. Some of audio's best tricks are simple and yet so deceptive." db

Concert Loudspeaker Processors

ANDREW T. MARTIN

Due to the ever-increasing public awareness of sound system fidelity, the pro-user's desire for dependable and "idiot-proof" audio equipment, and the loudspeaker manufacturer's constant battle to be one step ahead of the competition, the age of the concert loudspeaker processor has arrived.

Apogee Sound, Community Light and Sound, Eastern Acoustic Works, Electro-Voice, Meyer Sound Laboratories, Professional Audio Systems, and Renkus-Heinz are but a few of the companies presently manufacturing processor-based concert loudspeaker systems. Each company has their own design philosophies and individual product features, but as a whole, processor-based concert loudspeaker systems share the same general objectives. This article will attempt to explicate the most common design objectives and lightly touch upon the methodology in which the manufacturers are attempting to achieve the desired results.

The concert loudspeaker processor, in many instances, is nothing more than a combined electronic cross-over, signal delay, speaker protector, and response corrector packaged together for a particular loudspeaker and placed inside a single rack-mountable box. The objective of the loudspeaker processor is to optimize the acoustic output of the loudspeaker while protecting the loudspeaker componentry, all the while trying to remain as transparent as possible throughout the dynamic range of the loudspeaker. It seems simple enough, but applying this philosophy is extremely difficult. However, aside from the difficulties in design, the manufacturers are developing loudspeaker systems with definite advantages directly resulting from the loudspeaker processing.

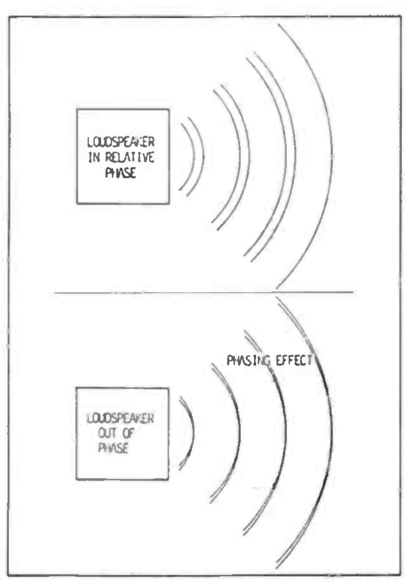
Aside from the obvious size and convenience qualities of the processor, there are a couple of other benefits as well. For example, by placing all of the working components in one box, and applying the same design philosophy throughout the entire unit, the processor is apt to be much quieter than individual components strapped together. As an aside, the technician's time can be spent in other places rather than patching together outboard components while trying to figure out input and output impedances. An additional benefit is the manufacturer's ability to customize the processor to a particular loudspeaker. Herein lies the real power of the concert loudspeaker processor.

By designing a loudspeaker processor for a specific loudspeaker, the manufacturer is able to incorporate circuitry for a known sound source to achieve optimal performance from that sound source as well as protect the sound source with the use of protective circuitry. The net effect is a seemingly more efficient loudspeaker system than traditional loudspeaker systems. As mentioned previously, the majority of the processed loudspeaker systems perform these optimizational functions with the use of electronic cross-overs, signal delays, speaker protection circuits, and equalization or speaker response correction circuits. The next four sections will touch upon these four functions of the loudspeaker processor.

ELECTRONIC CROSS-OVERS

The electronic cross-over in loudspeaker processors is much the same as other electronic cross-overs. The difference lies in the manufacturer's ability to design cross-over frequencies, slopes, and phase to meet the needs of the loudspeaker in order to create the optimum performance. This obviously cannot be accurately achieved with off-the-shelf cross-overs since the manufacturer has no way of knowing with what sound sources the cross-over will be in line.

Figure 1. A loudspeaker in or out of phase.



The loudspeaker processor facilitates precise tailoring of the adjoining cross-over frequencies to the specific sound sources they are serving, thus creating a smoother transition from sound source to sound source assuming the speaker components are chosen properly. The smoother transition gives the loudspeaker the illusion of being only one sound source rather than multiple sound sources working together, again assuming the speaker components are chosen correctly.

Cross-over slopes can be designed to make the transition from sound source to sound source longer or shorter, depending on the response output of the loudspeaker as a whole. Combinations of different cross-over slopes are also used to generate the desired results from the loudspeaker components.

Finally, phase can be adjusted for each cross-over frequency in order to align the acoustic wavefronts of the transitional sound sources. Alignment is achieved by matching the acoustic phase of the first sound source with the acoustic phase of the second sound source at the cross-over point, thereby putting the two sound sources in the same relative phase (*Figure 1*). Acoustic wavefront alignment is important in order to reduce the cancellation and summation of the acoustic energy being emitted from the loudspeaker. It is important to note that two sound sources with different coverage patterns, although in relative phase on-axis, will not remain in relative phase throughout their off-axis coverage pattern due to the difference in flare rate between the two sound sources. The proper choice of loudspeaker components is imperative for a successful loudspeaker, even with a loudspeaker processor.

Figure 2. Signal synchronization.

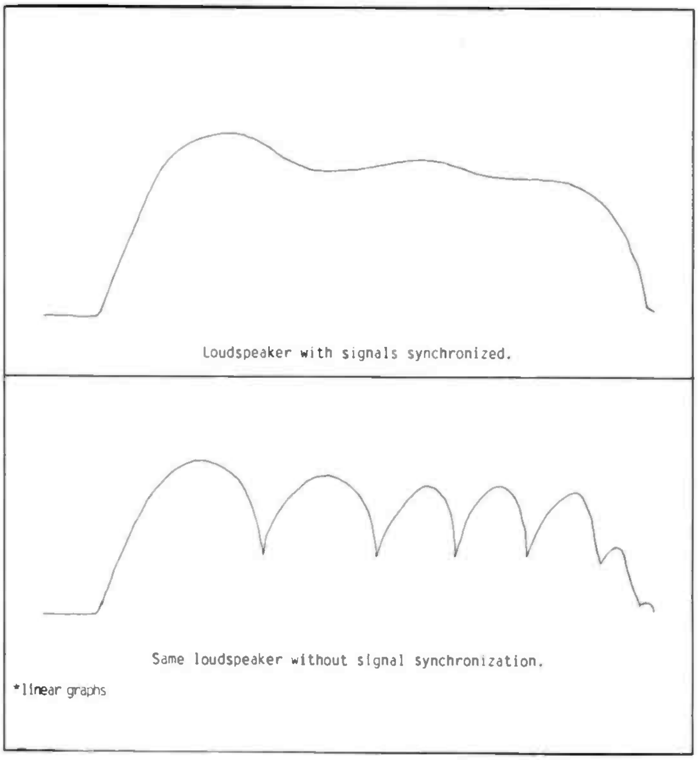
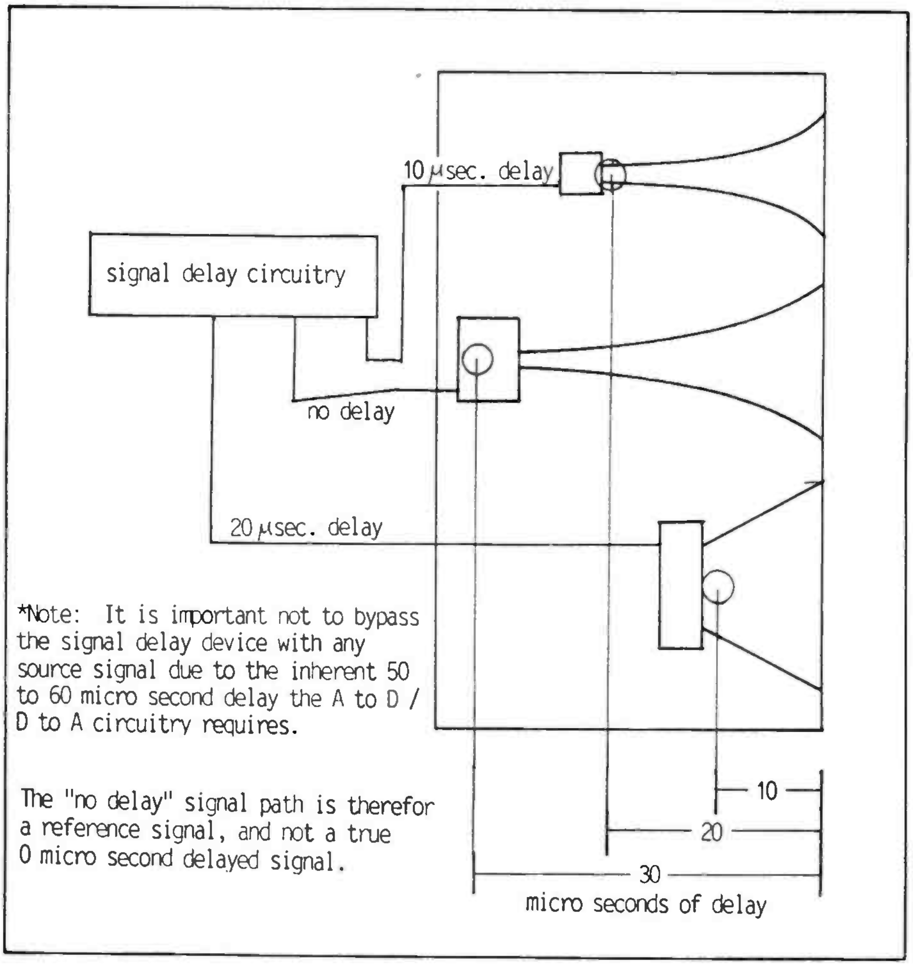


Figure 3. Signal delay.



Signal delays are utilized for sound source signal synchronization, a vital function in processors if the processor is servicing a loudspeaker with non-aligned sound source acoustic centers.

The acoustic center of a sound source is the point at which the acoustic energy appears to originate, and this point does not have to be in the center of the magnet assembly or in the throat of a horn. If the acoustic center of one sound source is not in physical alignment with another sound source, unnatural filtering effects will occur between the acoustic outputs of the sound sources. The filtering is commonly known as comb filtering (*Figure 2*).

Comb filters, when sizeable, will cause an audible distortion to the program material, and can be easily detected. However, comb filtering often will not be heard directly from the loudspeaker. Instead, the resultant of comb filtering will be heard in the environment surrounding the loudspeaker due to the adverse comb filtering effects on a loudspeaker's coverage pattern. A loudspeaker with sound sources that are not signal synchronized can emit very concentrated lobes in odd directions. Not only does this make loudspeaker placement difficult, due to feedback control, but it will also cause an increase in the reverberant field resulting from unwanted room reflections—thereby decreasing the direct to reverberant sound and decreasing intelligibility.

Since, as mentioned earlier, the acoustic wavefronts of the sound sources must be in alignment, and now the acoustic centers of the sound sources must also be in alignment, the manufacturer of the loudspeaker system must make a few decisions. Most manufacturers will physically align the acoustic wavefronts, and then introduce signal delays in order to synchronize the signals of the loudspeaker sound sources (*Figure 3*). Other manufacturers will develop an integrated sound source array that physically aligns both the acoustic wavefronts and acoustic centers, thereby eliminating the need for signal delay circuits inside the loudspeaker processor.

SPEAKER PROTECTION

The most common form of speaker protection in loudspeaker processors is achieved with limiting. The limiter will "squash" the dynamics of the program material, through the use of a very high compression ratio, should the limiter be switched on. The manufacturer can design the limiter in such a way that the switching circuitry and compression ratio are suitable for the speaker components inside the loudspeaker, and the dynamic range of the loudspeaker is maintained with the highest level of processor transparency. The method of triggering the limiting circuitry varies from manufacturer to manufacturer.

Some of the loudspeaker processors contain "sensing circuitry" that will constantly monitor the output of the amplification device and begin limiting when the amplifier goes into clipping. A very high-quality amplifier with the manufacturer's suggested power rating is necessary for this type of protection system to be of benefit. Other loudspeaker processors will incorporate a threshold gate that triggers the limiter when the program material exceeds the pre-set threshold.

Another form of speaker protection commonly found in loudspeaker processors is thermal protection. Thermal protection circuits are usually a form of low to mid ratio compression. As with the limiting function, the thermal protection is triggered in much the same manner. Additionally, there are some manufacturers that use "sliding crossover" functions in their loudspeaker processors to protect against thermal overload. This function changes the cross-over frequency of the sound source in danger until it is able to safely operate in the wider bandwidth again.

SPEAKER RESPONSE CORRECTION

Speaker response correction, or equalization, is incorporated into most loudspeaker processors. The processor package gives the manufacturer the opportunity to fine tune the components of the loudspeaker with great accuracy to achieve the desired loudspeaker output response. Some loudspeaker processors offer switchable and/or adjustable equalization functions for use when the loudspeaker is arrayed or placed in acoustically coupling environments.

Additionally, some loudspeaker processor manufacturers include "feedback suppression" circuitry with their processors. When the processor senses a feedback loop, the proper attenuation is applied until the feedback loop is repressed.

The age of the loudspeaker processor has enabled the sound system engineer to purchase a loudspeaker system with good performance, compact dimensions, and consistent acoustical characteristics: the three ingredients needed for successful performance. The processor-based loudspeaker systems have made it possible to travel from venue to venue and have an excellent idea as to how the sound system will respond at each site, while the time needed for set-up is, in most instances, greatly reduced. The loudspeaker system's owner has the piece of mind that the loudspeakers have a high resistance to destruction due to misuse. However, there are certain drawbacks to the processor-based loudspeaker systems as well.

The loudspeaker processor must be used in conjunction with the specified loudspeaker. Hooking up a processor to the wrong loudspeaker would probably not sound very pleasant, and may destroy the loudspeaker, although there are some manufacturers with "generic" limiter-based processing units that are capable of operating with other manufacturer's products.

The loudspeaker processor also has a tendency to distort the tonal values of the program material when the protection circuitry is employed. However, the tonal variation due to the protection function is usually much less detectable and preferred over the harsh distortion of over-excursion.

And finally, the purchase price of processor-based loudspeaker systems is extremely high when compared to conventional loudspeaker systems. As for the value—that is for the individual to decide.

Manufacturers Mentioned:

Apogee Sound, Inc., CA (707) 778-8887, Community Light & Sound, Inc., PA (215) 876-3400, Eastern Acoustic Works, Inc., MA (617) 620-1478, Electro-Voice, Inc., MI (616) 695-6831, Meyer Sound Laboratories, Inc., CA (415) 486-1166, Professional Audio Systems, Inc., CA (213) 534-3570, Renkus-Heinz, Inc., CA (714) 250-1035

A special thank you to Ken DeLoria, Apogee Sound, Inc.

Thank you, too:

John Wiggins, Community Light & Sound, Inc.

Ken Berger, Eastern Acoustic Works, Inc.

Joe Katowich, Electro-Voice, Inc.

Drew Daniels, JBL Professional

John Kirkland, Professional Audio Systems

Russ Farrell, Renkus-Heinz, Inc.

Gerry Tschetter, Yamaha Corporation of America

db

JOHN EARGLE

System Intelligibility Criteria

This article is Chapter 12 of the author's new book "Handbook of Sound System Design," now published by ELAR Publishing Co. Inc.

INTRODUCTION

The intelligibility of speech is of paramount importance in public meeting places and auditoriums, and sound reinforcement systems should always be designed with the goal in mind of increasing intelligibility over that afforded by unamplified speech. The most important factors in determining speech intelligibility are:

1. Speech level and signal-to-noise ratio. Speech can be understood over a wide range of levels; however, at lower levels, the signal-to-noise ratio must be of the order of 25-30 dB if speech is to be clearly understood. At higher speech levels, a lesser signal-to-noise ratio will often suffice.

2. Reverberation time. If the reverberation time in the 500 Hz to 2 kHz range is of the order of 1.5 seconds or less, then it will not decrease intelligibility. In fact, a reverberation time of 1.5 seconds or less is generally beneficial, since it increases the level of speech without interfering with the articulation of individual speech syllables. (The nature of the reverberation may be critical; in particular, strong discrete echoes will make for poor intelligibility if they are noticeably displaced from the direct speech sound source.)

3. Direct-to-reverberant ratio. For reverberation times in excess of 1.5 seconds, the overhang of sound tends to blur speech. In a sense, the reverberation behaves like a kind of noise signal, one that rises and falls with the level of speech itself.

4. Subjective considerations. There are both good and bad listeners and talkers. Given the same acoustical environment, an experienced talker will adjust his delivery to the conditions at hand, and he will be more clearly understood than an inexperienced talker. Likewise, an attentive listener with normal hearing will be at an advantage over an elderly person with some degree of hearing loss.

MEASUREMENT OF SPEECH INTELLIGIBILITY

The traditional measure here is syllabic articulation testing. In such tests, a reader calls out a list of unrelated monosyllabic words, and listeners at various parts in an auditorium write down the words as they perceive them. The articulation score is simply the percentage of syllables correctly identified. If a given listener correctly identifies 85 percent of the total number of syllables, then he will be able to understand normal speech in the testing environment with an accuracy of 97 percent or higher, due to the contextual nature of speech. If his articulation score is 75 percent, then he will be able to understand approximately 94 percent of the words in normal speech. This latter condition may be generally satisfactory, but some extra measure of attentiveness on the part of the listener may be necessary as well.

ESTIMATING SPEECH INTELLIGIBILITY

What the designer of sound reinforcement systems requires is a method of estimating speech intelligibility before a system is designed. Armed with such knowledge, he can determine just what kind of system may be best for a given environment.

Articulation Index (AI)

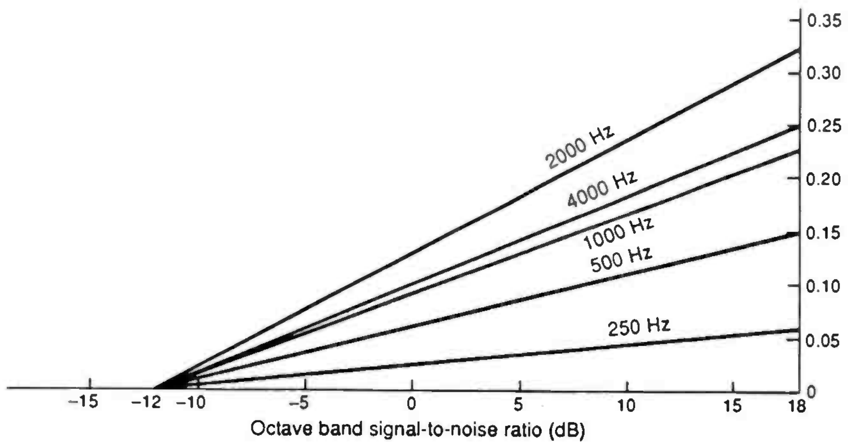


Figure 12-1. Calculations of the Articulation Index (AI).

Figure 12-2. Sample speech and noise spectra: AI calculation.

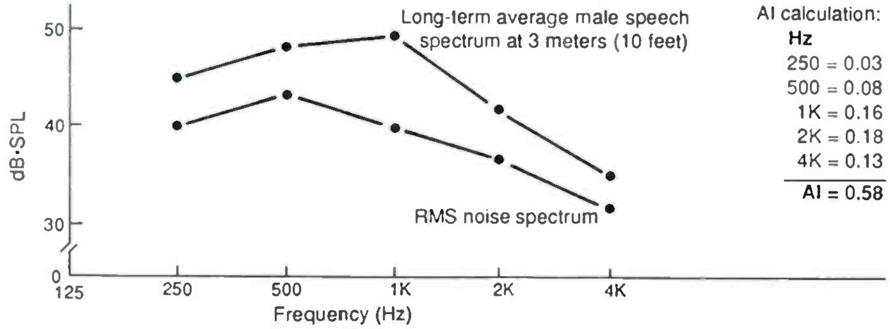


Figure 12-3. A comparison between AI and syllabic testing.

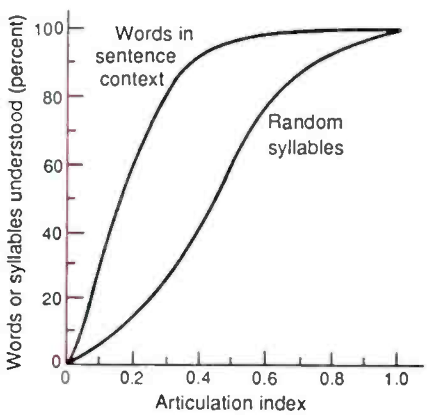
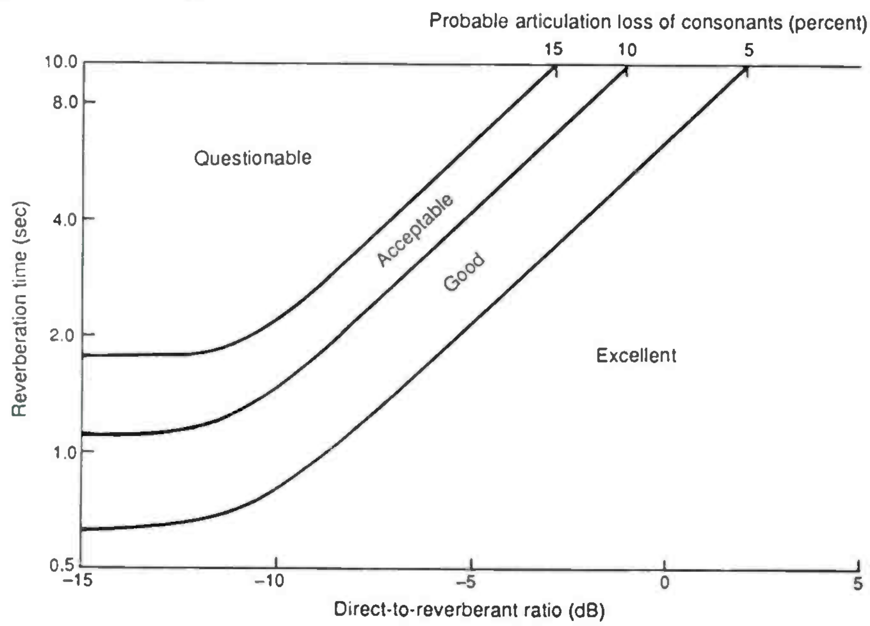


Figure 12-4. Articulation loss of consonants.



One of the earliest methods of determining the intelligibility of a transmission system is through the calculation of AI, as developed by French and Steinberg and later modified by Kryter. In calculating AI, the spectrum of ambient noise in the listening space must be measured, either on octave bands or one-third octave bands, over the range from 250 Hz to 4 kHz. Speech peaks are also measured over the same bands, and the two are compared at each frequency center. *Figure 12-1* shows a method for arriving at the contribution of each octave band in determining AI, as suggested by Smith. In using this graph, octave band RMS speech-to-noise ratios are measured and are individually weighted for their contribution to the AI over the normal 30-dB dynamic range of speech. Since it is easier to measure the RMS value of speech signals than peak values, it is this signal-to-noise ratio that is plotted in the graph. The assumption is made that peak levels of speech are some 2 dB higher than the RMS levels.

As an example of how to calculate AI, let us assume noise and RMS speech spectra as shown in *Figure 12-2*. Each band is individually observed,a nd its RMS signal-to-noise ratio entered into the graph of *Figure 12-1*. A line is drawn upward from this value until it intersects the diagonal line representing the octave band of interest. Then, a line is drawn to the right, intersecting the vertical axis. Each contribution, as measured on this axis, is noted, and they are summed to give the AI.

AI calculations are most applicable to non-reverberant conditions, such as those found in open-plan offices and many paging applications. There is generally excellent agreement between AI calculations and articulation tests made under these acoustical conditions.

Figure 12-3 compares AI with several measures of speech intelligi- bility. In general, an AI of 0.3 or greater indicates that general speech intelligibility will be adequate.

Peutz's Articulation Loss of Consonants

AI calculations have been found to give erroneous results if they are made under reverberant conditions. Peutz suggests a measurement of the articulation loss of consonants as a determinant of speech intelligibility which takes into account reverberation time, noise, and the direct-to-reverberant ratio in the frequency range of 1-2 kHz. can simplify Peutz's method considerably if we apply it only in those cases where the noise floor is sufficiently low (30 dB below speech peaks.) In this case, noise is not a factor in determining intelligibility, and Peutz's data can be replotted, as suggested by Augspurger, into the form shown in *Figure 12-4*.

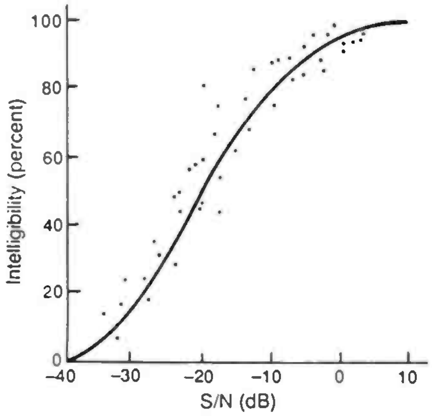


Figure 12-5. Signal-to-noise (S/N) vs. speech intelligibility; the curve gives the best fit to data points (after Smith).

In using this data, the reverberation time in the 1-2 kHz octave is measured or calculated, and the direct-to-reverberant ratios at various parts in an auditorium are measured or calculated. The data is entered into the graph, and an estimate can be quickly made of overall system intelligibility. Peutz states that this method has a limit in accuracy of about 10 percent, and it is for this reason that the data of *Figure 12-4* is broken down into only four broad zones of intelligibility.

The preceding methods are easy to work with, and they can be imple- mented while a sound reinforcement system is still on the drawing board if reasonable noise and reverberation time estimates can be made. There are two other methods of estimating speech intelligibility, but they require actual measurements on site. We will discuss them briefly.

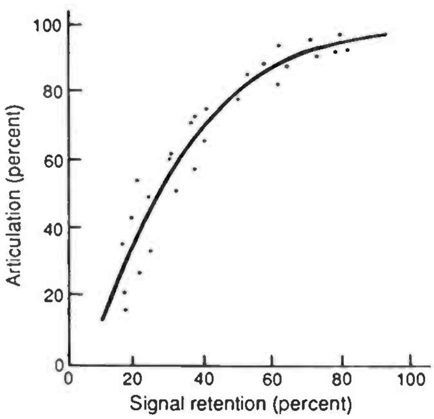


Figure 12-6. MTF vs. speech intelligibility: the curve gives the best fit to data points (after Smith).

Lochner and Burger's Signal-to-Noise Method

Lochner and Burger have determined that sound arriving within a certain interval after the receipt of direct sound is integrated by the ear and is useful. All sounds arriving after that time are considered as noise. The integration time is 95 msec, and the expression for useful sound energy is:

$$\text{Useful energy} = 10 \log \int_{t=0}^{95\text{ ms}} \alpha(p,t)\, p^2(t)\, dt$$

In this expression, $\underline{\alpha}$ is a fraction of delayed sound, integrated taking into a ccount the direct sound and the delay time. $p(t)$ is the instantaneous sound pressure.

The signal-to-noise ratio is defined as:

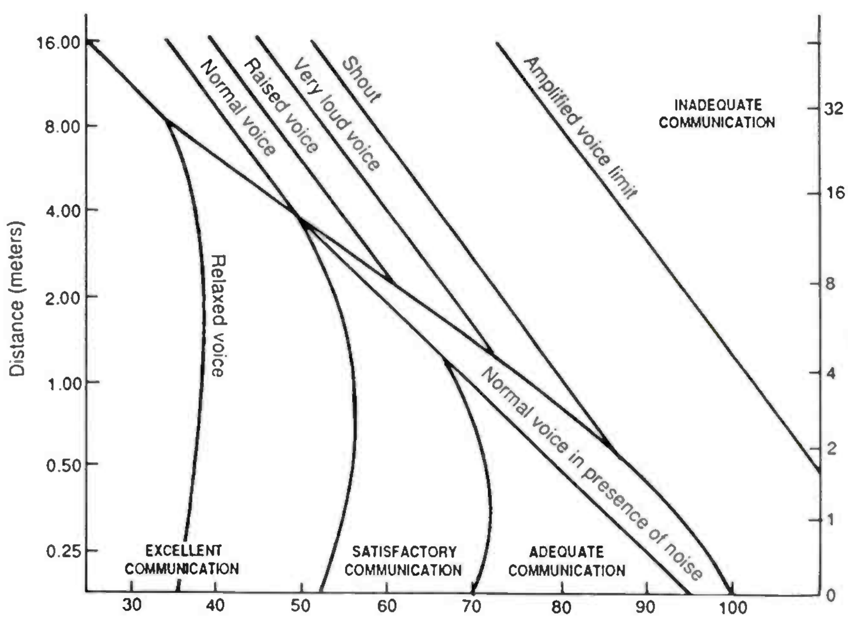
$$S/N = 10 \log \frac{\int_{0}^{95\text{ ms}} \alpha(p,t)\, p2(t)\, dt}{\int_{95\text{ ms}}^{\infty} p2(t)\, dt}$$

Figure 12-5 shows the relationship between the S/N calculation and speech intelligibility. Clearly, the method is an accurate one, but its implementation is quite complex.

Modulation Transfer Function (MTF)

Houtgast and Steeneken have proposed a method for measuring the effects which reverberation and noise have on signal integrity. Their method makes use of a test signal which is amplitude modulated and then reproduced in a room. The effects of reverberation and noise fill in portions of the modulation envelope, and the recovered signal can be compared with the original. In practice, a number of modulation rates are used at a number of one-third octave bands. *Figure 12-6* gives an indication of the accuracy of the method, stated by Houtgast and Steeneken to be within 10 percent. In

Figure 12-7. Permissible distances between talker and listener.



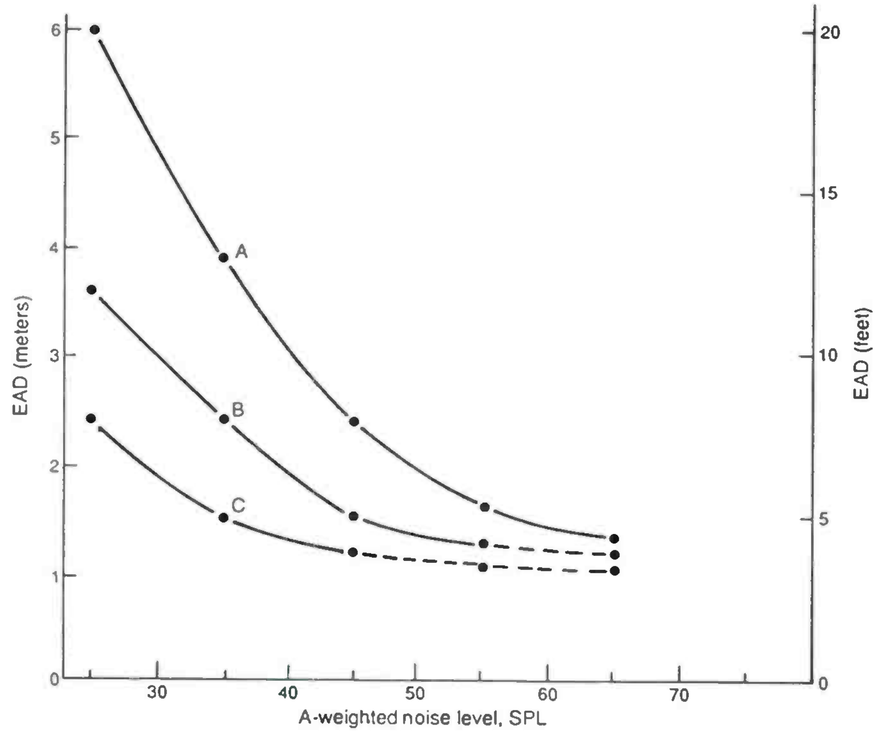


Figure 12-8. EAD vs. A-weighted noise level.

recent years, the RASTI (Rapid Speech Transmission Index) measurement method has evolved out of the work of Houtgast and Steeneken.

PERMISSIBLE TALKER-LISTENER DISTANCES FOR SPEECH COMMUNICATION

The graph shown in *Figure 12-7* shows permissible talker-listener distances as a function of noise level. The assumption is made that the listener and talker are not facing each other. If they are facing each other, then the noise level may be increased by 5 dB.

This graph clearly indicates the tendency of talkers to raise their speech level in the presence of noise, and continued communication under elevated noise conditions is a strain on both talker and listener.

As an aid in determining operating levels for sound reinforcement systems, the data of *Figure 12-8* will be useful. Here, we have plotted workable speech levels, as a function of noise level, for a raised voice (Curve A), a normal voice (Curve B), and a lowered voice (Curve C). In plotting these curves, the peak speech levels at a distance of one meter for the three types of talkers were set at 70 dB, 65 dB, and 60 dB, respectively. At lower noise levels, a 25 to 30 dB signal-to-noise ratio is maintained, while at higher noise levels the signal-to-noise ratio is allowed to decrease.

This data will be useful in establishing a value for EAD, as discussed in Chapter 10. db

BRIAN BATTLES

Ad Ventures

• You don't have decent production facilities at your studio? You're hampered by the inability to lay down those fancy spots like the folks in the big city? You've run out of ideas for creating new sounds with the primitive gadgets on hand? Many recording studios and radio stations are outfitted with ramshackle, abused equipment. You may have yet to get your hands on digital tape decks, high-quality delay units, special effects boxes, and other outlandish audio playthings that would let you run wild with exciting new sounds. But take heart: I produced a multi-layered, effect-filled extravaganza that bagged me a Clio Finalist certificate when I worked at a radio station that had a decrepit stereo production studio. The place was equipped with no more than a pair of Scully reel-to-reels, two BE cart machines, and a tired old Gates Executive mixing console.

You see, at that time I didn't know any real people that were using better equipment. To me, all the fancy stuff was just pictures in trade magazines. I naively reckoned that if my fellow helots were cranking out intricate *coups de maitre* with similar gear, then I could do it, too. You've seen the cartoons when somebody steps off a cliff and keeps right on walking, suspended in mid-air until some spoilsport exclaims, "Hey! You can't do that," causing the unfortunate slob to plummet to his violent doom. I was the one out there hovering, but nobody had told me what I couldn't do yet. So, I set out to emulate that **big station** sound, and I'll share some of the ideas I "invented." I hope to spark your imagination and help you avoid re-inventing the wheel.

PRINCIPLE OF CLEANLINESS

First, a basic principle of audio production: The importance of cleanliness in the studio can't be overstated. Electronic devices are vulnerable to damage from airborne dust, soot, smoke, as well as coffee, oils from human skin, food remnants, and many other common substances. It's critical for you to keep it as free of contamination as possible. This means 1) Don't eat around the equipment, 2) Wash your hands often, particularly after eating (and especially before editing tape), and 3) Don't place coffee mugs, soda pop cans, paper cups, beer bottles, or other beverage containers where they can be slopped on equipment. Sure, you may set a cup down "just for a second," but it only takes one spill to short out a circuit, coat a component with liquid, or allow a corrosive ingredient like sugar or salt get on electronic parts where it can slowly eat away at chips, solder, and wires.

Clean tape recorder heads before and after each session. This helps assure that if some lazy blockhead comes along and neglects their cleaning duties, you'll have covered it. Also, if you're ever in a rush (it's been known to happen), perhaps the serf ahead of you will have cleaned the machines upon finishing.

Since recording tape is simply a strip of plastic coated with billions of fine magnetic particles, it's logical to expect that each pass through the head block area induces a small magnetic charge on any metal parts in the vicinity. Magnetized heads and guides act like a weak version of the deck's erase head—when a tape goes by them, they tend to scramble some of the information on the tape. Since the higher frequencies jam more tiny signals on a given section of tape, they are the most fragile, and thus the easiest to destroy. If you never demagnetize your tape decks' heads, you will erase a few decibels of audio from your tapes each time you play them. It's easy to prevent this.

Demagnetize the tape heads, guides, and metal parts in the tape path on each deck at the beginning of each day or critical session. It only takes a moment to grab the hand-held head degausser and wipe away the built-up flux that can erase your tapes' high frequencies.

(Just be sure to clear the area of any recorded reels or carts first.) Always do the same thing to the capstan, guides, screws, or posts that also come in contact with the tape as it runs through the deck.

EFFECTS TECHNIQUES

On to special effects. There are plenty of home-grown techniques you can use to create a slew of interesting sounds when you produce a radio commercial.

You can use a compressor/limiter to create special effects, such as simulating the between-voice static heard on two-way radio conversations, or to make your breath sound like Darth Vader's. Play with it until you know how it works and what settings sound best.

You can also use EQ to create special effects by shaping the sound to simulate a tinny telephone conversation, a television speaker, or perhaps to imitate a robot voice.

Although your station may normally air commercials in stereo for the sake of hi-fi sound, you can take advantage of tape recorders that have two or more channels to create interesting production effects. Prepare a script for two characters and take both parts yourself, reading one on channel 1 (leaving appropriate pauses), then recording yourself on channel 2 as you listen to channel 1 on headphones, and carrying on a two-way conversation with yourself. This is very easy to do if your tape deck has a "sync" switch for playback so you can reproduce the first channel's signal right from the record head, since listening to the playback head introduces a momentary delay that throws off the timing of the second voice. To sound natural, practice leaving pauses that are of just the right length so you can slightly "step on" the first voice, meaning that you have voice #2 begin each line a split second before voice #1 finishes. This sounds more like normal conversation. For a high-energy delivery, read a bit faster and step on the voice a little more. This can be effective if you read in the same tone of voice on both tracks.

To sound like two different people, there are a couple of tricks you can try. If your tape recorder has a variable speed adjustment or pitch control, you might also record voice #1 or voice #2 (or both) while running the tape recorder at slightly different speeds. In addition, try reading in different voices or playing silly characters to produce a skit. Mix the finished two-channel tape down onto a mono track on another machine while adding sound effects or music to the background.

You can generate echo effects by turning up a tape machine's playback pot and feeding its output back into the mixing board's inputs. The original signal from your mic or other source first goes onto the tape via the record head, then is picked up a split second later by the playback head. The slight interval between the moment the tape travels from the record head to the reproduce head defines the length of time between repeats. Turning up the tape playback pot too far, however, causes echo that gets too loud, distorted, or tinny.

An interesting variation is to use "pre-echo." This is done by recording on one tape, flipping the tape over to play it backward, and dubbing it to another tape deck that you use to make echoes. When you flip over this second tape and play it back so the recorded sounds are again going forward, the echo now precedes the sounds.

You can generate another extraordinary tape effect by recording your voice normally, flipping the tape over, playing it back in reverse, and recording yourself on a second tape deck as you phonetically try to duplicate the gibberish your voice renders on the backward tape. If you then turn the second tape over and play it backward, you'll hear the words come out forward, and reasonably recognizable, but with an eerie, sort of otherworldly flavor. You can refine it with practice and if you desire, combine it with EQ, echo, pre-echo, and other modifications to create a strange sound that can suggest aliens, ghosts, or computers. Experiment by reading in a robot-like monotone, spooky moan, foreign accent, or some other vocal variation.

Playing with tape speed is fun, too. Most studio tape decks today have some kind of variable pitch control built-in, but you can get unique results by wrapping your capstan with masking tape. Cover it with one perfectly smooth layer and you get a fairly steady speed. Make it warble by wrapping it one-and-a-half or one-and-a-third turns. Try it on recording, playback, or both. Fool around a bit, but be sure to clean the adhesive off the capstan when you're done.

These only scratch the surface of the wide variety of fascinating production techniques that you can use to produce great radio commercials on a shoestring budget and using the most basic of recording equipment. From electronic cottage or backwoods radio station to world-famous recording or big-city broadcast center anyone can get exceptional results if the people who operate the equipment have learned to improvise, experiment, and push their technology to the limit. Your imagination is the most powerful audio production tool in the studio.

BEHIND THE SCENES

As I write this installment of *Ad Ventures*, I'm sitting at my rickety desk surrounded by dozens of cardboard boxes containing papers, office supplies, electric and electronic gizmos, mechanical gadgets, certificates, photographs, clothing, garbage, and even some food. This miserable accumulation of rubbish constitutes the majority of what one might call my material fortune and worldly goods.

For reasons too complex to recount in this space, I have migrated again, this time from Boulder County, Colorado, mountainous home of granola-munching college students and schussing yuppies, to a quiet little rural town of 10,000 in southeastern Connecticut. Having left CareerTrack Publications to form Porkpie Productions, I am now a full-time writer, commercial producer, and entrepreneur.

Comments and suggestions from you are what keep this column alive, so please write to me at P.O. Box 176, Colchester, CT 06415 or in care of db, 203 Commack Rd, Suite 1010, Commack, NY 11725.

And now, the moment you've all been waiting for...

PROMOTIONAL MANEUVER NUMBER ONE

It's here! Hot off the duplicating machines are the first copies of my long-promised audio tape program, *How To Produce Great Radio Commercials*. It's a four-cassette reference chock-full of advice, techniques, ideas, and tricks of the trade for anybody who wants to learn more about putting together profitable and award-winning radio spots. You also get to hear samples of my work, commercial parodies, and some bonuses. Not sold in stores! Order your copy by sending check or money order for $99.95 (in U.S. funds) to Porkpie Productions, P.O. Box 176, Colchester, CT 06415. I'll provide free domestic shipping. db

ROBYN GATELY

The Sound Contracting Engineer

Pianos I Have Known

• Of all the instruments known, none elicits more opinions than the art of mic'ing a piano. I have never seen more opposite view points expressed by more people, outside of politics. One person's idea of perfection can be another's idea of a nightmare.

One of the problems in discussing piano mic'ing, and the advantages and disadvantages, is that there is no common consensus on what a piano really sounds like. This is caused by the fact that there is no agreement on where the listener should be seated when listening to the piano.

Some people prefer the sound when the piano is located on the stage and they are sitting in the third to seventh row of the theater; some people prefer to feel as if they are sitting, playing the piano themselves; while a third group likes the feeling of standing in the curve of the piano with the top open, watching the player. Obviously, these three very diverse opinions on how the piano should sound in the hall leads to very unique concepts on how to translate a piano onto tape or into a PA. (Never mind the additional effects of piano top open, short stick or closed.)

DIFFERENT POSITIONS

Sitting in the audience creates the sound of a piano with the natural reverberance of the hall (sometimes described as being "airy"). However, this listening position also is very muted on the sound of the hammer striking the strings (which creates much of the transient of the note). The piano players position has much stronger transients due to the players proximity to the hammers; while the third position has less transients but a more full bodied sound due to the more direct reflections off the piano lid.

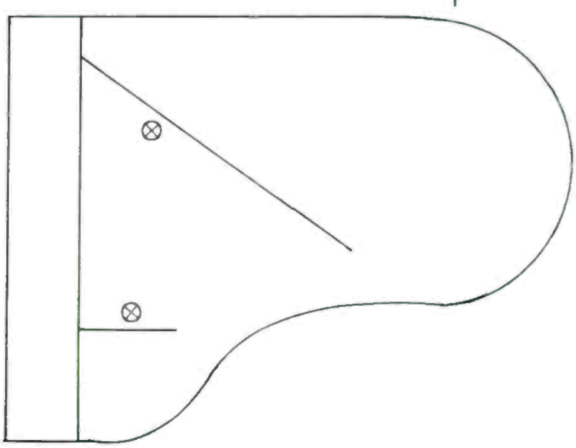


Figure 1. Mic'ing a piano using two microphones. The primary mic is over the strings about 2-in. away from the main strut and 6-in. away from the hammers. The second mic is mixed in to preference, if at all. Realize that the use of the second mic also brings phase cancellation.

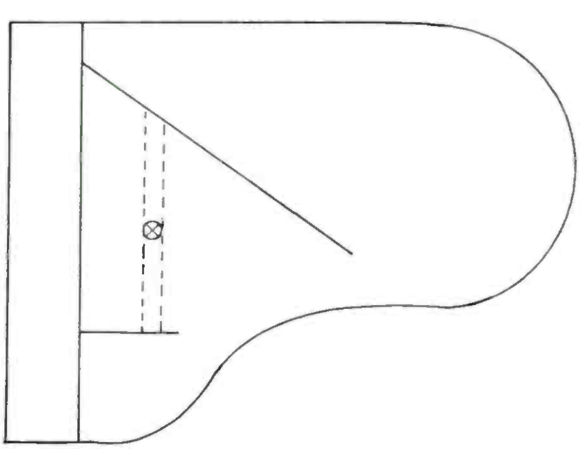


Figure 2. Duct tape across the two struts with the mic laying on top of the tape, in the middle.

The ability of the lid to be closed, open on a short stick, or open on a long stick confuses the issue even more. (One act I work with even carries her own mid-size stick). My ideal position is with the top open (long stick) sitting with the piano player on the bench, but that's just my opinion.

The invention of the piano top was a great volume control from the days when changing acoustics was the only volume control. For quiet work, one would just close the lid, providing at least 10 db of attention (volume drop) and a significant muting of the transients. Opening the top allows the volume and transients their greatest ability to fill the music halls of the day, while providing an increase in the deep bass. Of course, the short stick is a compromise between the two.

The problem in modern times is that everyone in the music halls want to hear the piano just like it was in his living room. This has necessitated the use of sound systems to augment its volume. Of course, the first problem is deciding which piano sound you would like to present to the public; and secondly, how do you deal with the problem of leakage from other instruments.

Because of my position in sound reinforcement, I have had the opportunity to see a wide variety of techniques used to achieve the various desired results. In this article I will present about a dozen different ways to mic a piano. However, it should be realized the sound of one type of mic'ing can be altered to sound closer to another type of mic'ing through the use of the tone controls, especially the treble control.

HOT SPOTS

My personal preference is the result of being taken for a "drive" around the piano soundboard many years ago. With the piano top off, Neil Shurmur, Sammy Davis Jr.'s soundman showed me a number of hot spots on the piano.

The loudest and fullest sound on the soundboard seemed to come about 6 inches down from the hammers, along the long strut in the piano frame. Actually, the loudest point in this hot spot was about 2 inches away from the strut towards the middle of the keyboard.

Because of this "discovery" I elect to mic the piano with one or two Lavalier type omni directional microphones. (*Figure 1*) Now, I already know that half the soundmen reading this article are saying a million statements about my sanity right now, but think of the following reasoning: First, an omnidirectional mic, dollar for dollar tends to be flatter in response than a directional mic. Secondly, if you place directional microphone in the piano at this spot, even with the piano lid wide open, there still is a reflective surface within about 5 inches of the back of the mic (seriously damaging both its rejection pattern and frequency response). Third, psycho-acoustically, what you don't see, you don't pay as much attention to.

I mic the piano with one lavalier taped to the main strut, 6 inches away from the hammers. I placed the mic with the element on the inside wall of the strut, hanging there off its wire, with the wire taped coming down the strut to the foot of the instrument. The cartridge to the mic is left in the body of the piano, while the mic wire is then led around to the back of the instrument and draped down the leg. The audience does NOT see a jungle of metal around the piano, but rather, the instrument in its elegant natural state.

The second mic, if used, is a lavalier taped to the inside of the short strut at the 6 inches from the hammer position. The point of this mic is mostly as a spare, but is sometimes handy when a piano is a "dog" with poor high end.

When utilizing the lavalier mic'ing technique, I usually end up adding some highs on the channel EQ for the main mic, and turning down some of the lows. This piano technique has been used in all kinds of situations from classical to rock. In fact, one well known pianist bought the mics from me on the spot. But, it's not the mics as much as it is the position.

Despite all efforts to reduce leakage into the piano, I've found that the best results come from positioning the loud instruments away from the piano. It becomes very rough to mix when you have to balance the rest of your mix against the leakage into the piano. Despite many claims of various techniques for reducing the leakage, the problem still occurs because all sound is able to resonate through the soundboard. It can not differentiate between sound generated by the string or by a rock band. Piano pick-ups that take sound off of the soundboard obviously must deal with the realities of this situation, but, in addition, lose the sound of the transients (hammer strike) due to the nature of their use.

Figure 3. A mic placed at the foot of the piano for the low end, and one positioned in the center of the struts for mid range. This is also a preferred mic'ing technique used by many sound mixers.

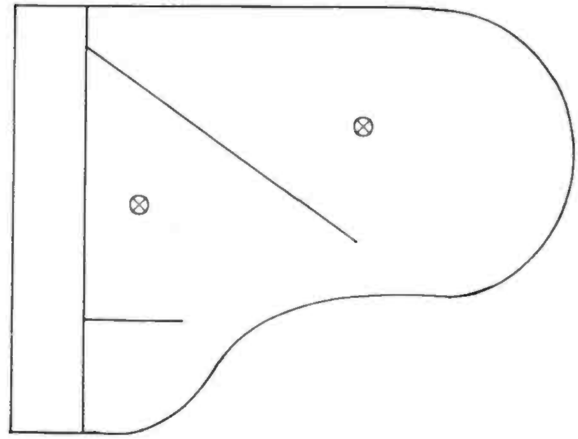
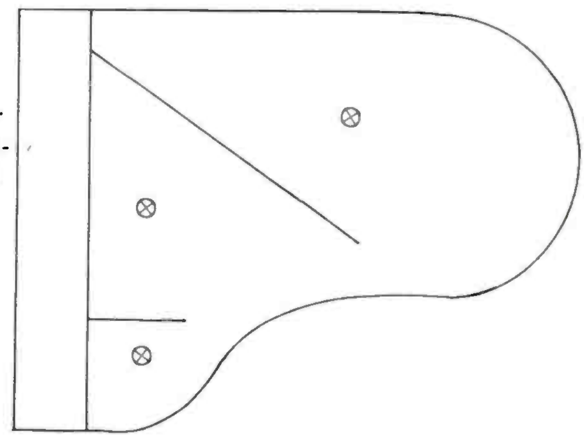


Figure 4. Three microphones, one at the foot (bass), one in the mid-range section, and one for the high end. This configuration guarantees that all possible sounds of the piano are brought forth into the mix.

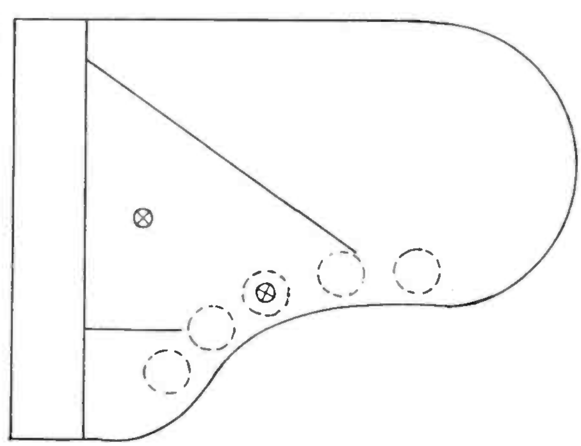


Of course, another method of mic'ing is to place one or two mics on stands in front of the piano. Advantages can be the directional nature of the mic, and its ability to be placed the 2 inches away from the main strut. However, I feel that the disadvantages outweigh the advantages for the reasons previously stated.

I have seen some people take a large piece of duct tape and stretch it across the two struts, and then rest a mic in the middle of the duct tape (*Figure 2*). The point of this is to get the sound from the middle of the piano, while obtaining the un-mic'd "look."

The third figure shows another popular mic'ing technique, the use of one mic to catch all if the string action, while a second mic is placed over the

Figure 5. Placing a mic in a hole near the curve of the piano emphasizes the piano sound in the mid range and reduces the amount of the hammer transients. This technique requires some judicious use of eq., and can be helped by another mic over the hammers.



middle of the vibrating bass strings. The point of the first mic is similar to the procedure in *Figure 2*, however the purpose of the bass string mic is to receive the sound at the point in the string that is loudest. A vibrating string is loudest in the middle, and this results in some people mic'ing the bass strings according to their vibration instead of their interaction with the soundboard.

A variation on this technique is utilizing three mics on the piano, the first two mics as above, with an additional mic for the high strings (*Figure 4*).

One useful parameter in two or more mics on the piano is the ability to change the PHASE of one of the mics. This change causes the midrange to become more prominent or reduced. Of course the use of more than one mic on anything causes phase related cancellation, which is why my preference is for one mic properly placed on the instrument.

Placing the mic in one of the holes in the curve of the body (sometimes called the "sound holes") is a fairly common technique for people who prefer a more "airy" sound. Whether this mic is laid into the piano or placed on a stand, the sound strived for is usually one of reduced transients and a more "full-bodied" sound.

MORE OPTIONS

In using non-technical terms through the years, I've discovered that certain words have meanings that are associated with certain phenomenon. For instance, "bright" means treble, so "dark" means bass; "warm" usually refers to midrange, while "full body" means an emphasis on bass and midrange over the treble. Of course, adding a mic over the strings to the mic in the hole allows you to adjust the amount of transients from the hammers to your "airy" sound.

I have also seen these other techniques: a mic placed outside of the piano at the curve halfway in height between the strings and the lid, and also a mic placed at the foot of the piano (outside) so that the mic picked up what the conductor hears. Neither have a lot of relevance to sound work.

Another legitimate mic'ing technique is placing PZM type mics taped to the piano lid in the positions of *Figure 1* and *Figure 3*. This produces what many people feel is a very acceptable piano sound, however just as many people don't like the sound of these types of transducers. A recent show with the Modern Jazz Quartet provided a different viewpoint; they like the positioning of the PZM users but not the sound, so they wanted standard mics placed in the PZM positions, with the head of the mic facing the lid so it captured the reflective sound of the piano lid while rejecting the direct sound.

I have also in the past seen some people take the piano with a mic underneath to capture just the sound from the soundboard, or in various other combinations. Although I do not prescribe to this theory for grand-type pianos, it can be very useful when dealing with uprights.

The point of presenting all of these ideas is to give you more options. Take a tour of the piano yourself, experiment, be creative & don't make it feedback! Remember, experiment on your own time, don't make your audience experiment with you. Make sure you take advantage of the EQ on your board channels to change what you've got. After all, nothing is ever perfect, so maybe you'll discover something better—just don't be afraid to try. db

RICK SHRIVER

A Minimalistic Philosophy

Rather frequently I am asked for advice on the subject of sound recording, perhaps because I was the first kid on my block to own a recording studio. Many of these seekers are novice recordists who are setting up their own studios or struggling to get "that big sound" from their small rigs. Having been in this business for about ten years, I have developed a framework for approaching this mystical art/craft/science that I willingly share.

"Minimalism" is the term that I have chosen to describe this methodology. According to Webster's, a minimalist is one who "advocates a program...of a minimal or conservative kind." It is akin to frugality, when frugality is defined as efficiency or not being wasteful, versus "cheap." (But, since the culinary world already has a Frugal Gourmet, I have chosen to be a minimalist. It will be less confusing.)

A LOGICAL APPROACH

The minimalist approach seems particularly logical for those of us practicing our craft at some level lower than the state-of-the-art. Beginners will find this paradigm invaluable. Further, minimalism is a philosophy that may be applied to many aspects of the recording field: equipment selection and purchase, production, engineering, and mixing, to name a few. Let us examine each of these areas in turn, from the perspective of the minimalist.

The rationale which we have long applied to equipment acquisition in our operation has been to purchase items which are of the required fidelity, regardless of other considerations. Implicit in this deceptively simple maxim are several things. Buy those pieces of equipment which are of professional quality, versus professional brand. If the required performance is available under a private label brand or through a consumer electronics store—buy it.

We have found, for example, some very good values at places like Radio Shack. Outlets such as this often market products which are produced for them by major manufacturers such as Koss, Sony, Crown, Fostex, Shure and others. This had led us to select headphones, microphones, pre-amplifiers, meters, monitor speakers and other items from these sources. Ultimately one pays a lower price because the brand name is not a conspicuous feature on the equipment, but the quality is still very high.

USING WHAT IS NEEDED

In our operation, we are also cautious of buying equipment that is actually higher in quality than we require. This almost blasphemous notion recognizes that we are, after all, a narrow gauge eight-track studio. Much of our product will be heard over the limited bandwidth medium of radio, or be reduced to an audio cassette for playback on consumer systems, while keeping future growth and expansion in mind.

Minimalism in engineering may take several forms. In laying tracks, resist the temptation to use more microphones than really are needed. I have noted on several occasions that neophyte engineers will instinctively regard the drum kit as an instrument requiring at least one microphone per striking surface—large kits demanding up to a dozen mics.

Experience has shown that is not necessarily the case. A simple set up placing microphones on the kick, snare and two overhead will often produce excellent results. Listen. Let your ears decide if that is really all that is required. Until one has a solid understanding of placement and phasing considerations, the caveat "less is best" should be kept in mind. After the basics are mastered should come the expansion and experimentation.

Minimalism should always prevail in equalization and the use of effects.

Just be sure that the frequencies which you will want to bring out of EQ in the mixing can replace what is not there. But, essentially, it is my belief that the emphasizing, boosting and cutting should go on in the mixdown. Good basic tracks will contain a full range of frequencies with which to work.

Of course, the "rule" stated above, like all others, is made to be broken. There are certainly times when severe equalization is required in the tracks to achieve the desired end result. Overall, though, try to capture the entire spectrum initially, alter it later.

Generally, effects should not call attention to themselves. Another common tendency for those new to engineering is to overuse certain effects, notably the application of reverberation. That is particularly true for vocals. Reverberation should only be used to add a sonic environment to the voice, a sense of the space in which the performance took place.

Begin with no effect on the signal, then very gradually bring it up in the mix, Stop when the vocal exhibits the desired ambience, before you can "hear" the reverb. Again, experimentation is the only teacher. and, as previously stated, this is obviously another rule made to be broken, but is a great help in learning to use effects.

Inasmuch as minimalism is the achieving of maximum results with minimum resources, studios with limited track capability will always benefit from this philosophy. Taking full advantage of the narrow gauge studio may require "pre-mixing" drum tracks, background vocal tracks, rhythm tracks or others.

A stereo drum mix can be created during the track laying and save many precious tracks. We generally place the drum mix on two tracks of the multitrack with the conventional stereo placement, snare and kick in the center and toms and cymbals panned across the stereo field.

Limited numbers of outboard devices may also necessitate recording of effects during the track laying, rather than during the mix. While this commitment to a given type and amount of effect will make many engineers uncomfortable, it will allow the minimalist engineer to take fullest advantage of the multi-effect units now popular.

Our practice is to put effects on the drum tracks as they are laid. Similarly, we will often do the same for guitar and background vocal tracks. This allows us to "save" the effect units for use during the mix for lead vocals, lead guitar, horns or other predominant tracks.

DON'T OVERMIX

Finally, minimalism in producing will demand a layered but uncluttered sonic environment created through the recording process. Again, many novices will exhibit a tendency to "overmix" tracks by adding too many parts to the piece.

As the mix is built, resist the temptation to bring in all the parts simultaneously. Often a more powerful cut emerges through the interplay of the basic rhythm tracks, rather than through the addition of many guitar parts, horn sections, or layers and layers of vocals. Bring parts up in the mix selectively.

As the instruments are juxtaposed in the sonic space of the stereo field, so should they be juxtaposed in frequency content. that is, a mix is structured in at least three domains: where the instrument is placed "left to right" (achieved through panning), where the instrument is placed "near to far" (achieved through level control), and where the instrument is placed in terms of "harmonic content."

This third element of the sonic space is accomplished through the layering of frequencies. an instrument whose harmonic content is lower frequencies is placed at the "bottom" of the mix. Usually that is the kick drum or bass guitar. The next instrument introduced into the mix should be one whose harmonic content is the next highest. If the lowest frequencies in the mixing are contained in the bass guitar or synth track, the next track should then be the kick drum, whose frequencies are slightly higher.

So, the mix is constructed of layers of frequencies, layers of levels and spatial positioning or panning. This will prevent signals from getting cluttered or lost in the mix because their levels, frequencies or position are to close together.

Keep in mind that what I've termed the minimalistic is approach is only one among the myriad philosophies which can be applied to the field of audio recording. Its particular usefulness is for those who have limited budgets, limited facilities or those who are beginning their careers as recordists. It would certainly appear to be just as applicable to many other practitioners. And, as stated above, minimalism has no "rules;" it is a framework for problem solving. db

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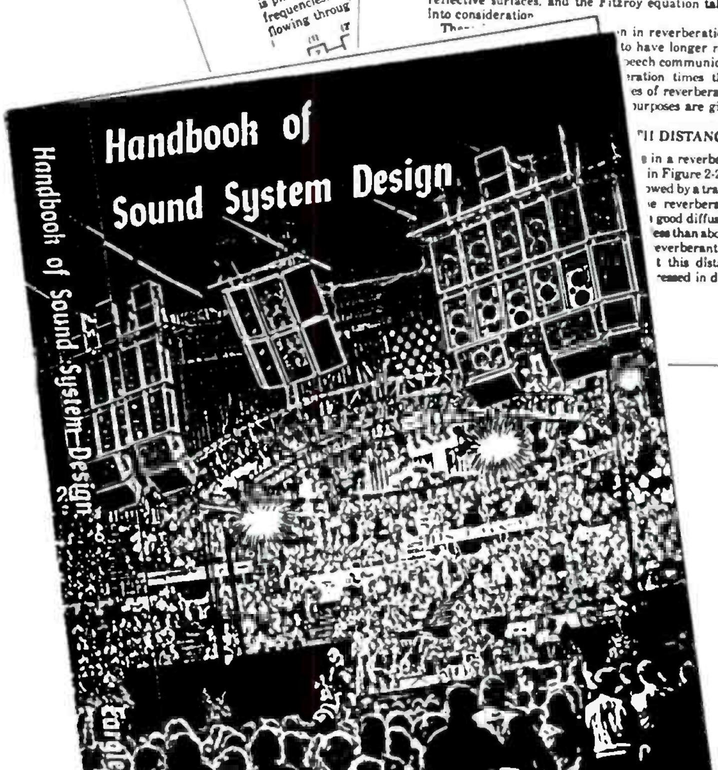

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Sound Reinforcement in Central America and the Caribbean

ED LEARNED

This issue is devoted to the Sound Reinforcement Scene: theory, layout and construction. Fledgling engineers will find these topics addressed in my articles: I've always tried to let my readers know the theory and logic behind my audio decisions. I've faced a lot of problems in my 15 years of independent engineering, and my experience is always taxed by the problems I encounter overseas. Read on: the solutions to overseas problems work at home, too! I'll be writing about North Africa and Micronesia in upcoming issues.

In Los Angeles during the late 1950s, a struggling saxophonist named Ornette Coleman was introduced to a nineteen-year-old trumpeter named Don Cherry. The two men hit it off, and in 1959 they recorded an album, *Something Else*, for the Contemporary label. The rest is history: jazz history. This record was seminal in the "free jazz" movement of the 60s; Don Cherry became a major voice in this new style. Throughout the 60s, Don worked with Sonny Rollins, John Coltrane, Albert Ayler, and Steve Lacy in addition to the trailblazing Ornette Coleman Quartet. He continued to expand his horizons throughout the 70s, touring the world and incorporating the music and instrumentation of many cultures into his musical identity. Don described his 80s music as "world folk music in jazz forms," eschewing the "avant-garde" label often hung on him.

> Author's Note:
>
> I hope that my faithful readers will enjoy this, my first international sound reinforcement article submitted to db in 1989. You may notice an overlap this month: I previously wrote about Central America in the July/August 1988 issue. My main reason for writing these articles is to provide my peers with information on acoustics, electricity, and equipment available in each region should they follow in my footsteps; I sure could have used such a reference when I started. We played venues on this tour of Central America that I didn't visit with Wayne Toups; hence, I included them in my article.

A NEW TOUR

Avid readers of **db** will recall that USIA (United States Information Agency) contacted me regarding Don's proposed tour while I was in Southeast Asia working with Benny Golson (see **db**, January/February 1989). I'd always enjoyed Don's music, and had previously worked with him on several occasions, so my immediate acceptance of this tour surprised no one! I was particularly intrigued by Don's current band. The group, a quartet, would consist of reed man Carlos Ward, Bob Stewart, tuba, Ed Blackwell, drums, and Don on pocket trumpet, keyboards, and doussin' gouni (Malinese "hunter guitar"). What, no bass or piano? This instrumentation was unusual for a jazz group, yet in a way very traditional: early New Orleans jazz groups were brass oriented, and used tuba instead of bass for the low foundation. I relished the challenge this group would present to a sound engineer. Our tour would take place between November 21 and December 20, 1987, a perfect time to escape my Michigan home for a warmer climate. Our itinerary included Nicaragua, El Salvador, Jamaica, Trinidad, the Dominican Republic, and Barbados.

When I returned to the U.S. in early October, I contacted Sandra Murphy, the USIA program officer handling Don's tour. My major concern, as always, was to ensure that quality sound equipment could be provided for our use. When I first began international touring in the late 70s, "state-of-the-art" sound systems overseas were the exception, not the rule. This situation changed during the 1980s. While there were still parts of the world I wouldn't tour without sound, it is possible to find enough quality equipment in most places to handle an acoustic jazz group such as the Don Cherry Quartet. My previous experience in the region led me to conclude that it would be possible to pick up sound locally. The one exception was Nicaragua. My tour of the country with Wayne Toups in early 1987 (see db, July/August 1988) illustrated the problems of procuring adequate PA equipment, and USIS-Nicaragua had reported no improvement in this situation. After talking things over with Don, Sandra and I decided to bring a small system to Nicaragua. When we left the country, this system would be returned to Washington; I would then rely on locally contracted sound for the duration of the tour.

USIA owned a complete sound system, using Bose 800 series PA speakers, Peavey monitors, and Crest amplifiers. The system was stored in Paris and used primarily for tours of Africa. With the scope of PA operations continually increasing, USIA found the need for more equipment, and purchased another complete system in the summer of 1987. This system was also slated for Paris storage, but prior to shipment Sandra snagged it and arranged for its transfer to Nicaragua, where we would make use of it. The new system was based on Bose 802 and 302 speakers, JBL monitors, and QSC amplifiers. A Soundcraft 200B console provided mixing, with Yamaha, dbx, and Shure components rounding out the house electronics complement. Peavey electric instrument amplifiers and a Yamaha PF-85 keyboard were included for band amplification. Don required a keyboard, so we arranged for it to accompany us on the entire tour. Bob Stewart required an amplifier for his tuba, but we decided to pick that up locally, along with the PA equipment, once we left Nicaragua. Microphones were another area of concern to me. Top quality mics are rare in some underdeveloped countries, and even the Shure SM-58 and SM-57 mics included with the USIA sound system are not noted for their sonic accuracy or mic-to-mic consistency. Maintaining consistency in group sound is of

Figure 1. Preparing for a sound check at the Edgar Munguia Theater.

paramount importance when touring anywhere, but especially overseas, where everything else changes radically! I wanted to ensure that my instrumental sounds would be consistent, so I arranged to carry my own microphones.

PLANNING AHEAD

With Nicaraguan concert planning complete, I turned my attention toward the rest of the schedule. I drafted a sound reinforcement and equipment rider outlining our needs. This covered front-of-house console and electronics, house and stage monitor speaker requirements, stage layout configuration, and stage needs such as a bass amp for Bob, a drum rig for Ed Blackwell, and music stands for each band member. I again used a watts-per-audience-capacity formula when describing PA system power parameters; I'd found this helped local organizers understand the need to provide larger systems for larger venues. I did not want to be underpowered. Sandra sent this off via cable to each USIS office we would visit post-Nicaragua. Response was excellent in most cases, so we had a pretty good idea of what to expect in each country before we left the U.S. I was pleased to discover that I would not have to worry about converting voltage: most of our destinations used a U.S. standard 120/240 volt electrical system.

I left for New York on November 20 to rendezvous with the group for our pre-tour briefing. Don had returned that afternoon from a European tour; in deference to his jet lag, USIA agency escort Toney Seabolt quickly conducted all necessary group business. After the short meeting, most of us stayed to discuss the environmental, cultural, and political climate of each country we would visit. We also discussed, with Francis Blackwell, Ed's special medical needs. Ed suffers from kidney failure, and requires peritoneal dialysis: simply put, he carries with him different bags of chemicals and, through tubes implanted in his body, gives himself his own dialysis. Without the need for hospital stays or dialysis machines, Ed is able to travel and perform. However, fluid changes must be made within a specific timetable, so we modified ground transportation, sound checks, and concert schedules with this in mind. We planned to carry about a hundred pounds of medical equipment and chemicals with us, about a ten-day supply; extra supplies would be shipped ahead to local U.S. embassies and "stockpiled" for us. Our schedule had been carefully crafted to allow this. I agreed to supervise the transportation of medical supplies and musical instruments we'd be carrying; this would be sent together as excess baggage. Experience dictated this approach: it is the best way to ensure that group equipment travels on the same flight with the group. It also allows personal attention during customs clearance, where delays could be costly to a tight schedule.

Our tour began on Saturday, November 21, as we all struggled to be at New York's LaGuardia Airport by 6:30am. Our Eastern flight took us to Miami, where we had a three-hour layover before connecting to a TACA flight bound for Managua, Nicaragua. We had tagged the baggage all the way through to Managua, but I remembered my previous experience transferring stuff to TACA in Miami; I told Toney to hold no truths self-evident. He'd reached the same conclusion, and relentlessly pursued TACA officials to ensure proper transfer of our bags. I was ushered into the baggage holding area to identify our stuff and confirm the transfer. This might seem excessive, but it was the only way to guarantee that our stuff would make the same flight. We were delayed an hour leaving Miami, and spent an additional hour on the ground in El Salvador waiting for a connecting aircraft, so we didn't arrive in Managua until just after 8pm. We were met by two familiar faces: CAS (Cultural Affairs Specialist) Pauline Frixione and CAO (Cultural Affairs Officer) Lois Mervyn. In deference to our tired musicians, Toney sent the group off to the hotel with Lois. The rest of us stayed to collect the baggage and handle customs clearance.

Many changes had occurred since my last trip to Nicaragua only five months ago. Everyone was now required to exchange U.S. $60.00 into cordobas (local currency), prior to customs clearance, at the official (read: financially unfriendly) exchange rate. Two things were apparent: design of the currency had changed, and inflation was even worse. My friend from the Ministry of Culture, Jose Benito, was not there to greet us; a new representative replaced him. I inquired discreetly as to his whereabouts, and was informed that he "was no longer employed." Things do change fast in the Nicaraguan government. Political tensions in the region had relaxed considerably, as evidenced by the way we rapidly cleared customs with only a perfunctory inspection of our documents and no search. Pauline brought me up-to-date on the latest happenings since my last visit, which included the arrival of Lois: she had been CAO in Honduras a few months before, and had recently been re-assigned to Managua. Once reunited at the hotel, we held a short briefing so Lois and Pauline could acquaint us with our local schedule. We were then free to unwind after our first long travel day—it wouldn't be the last.

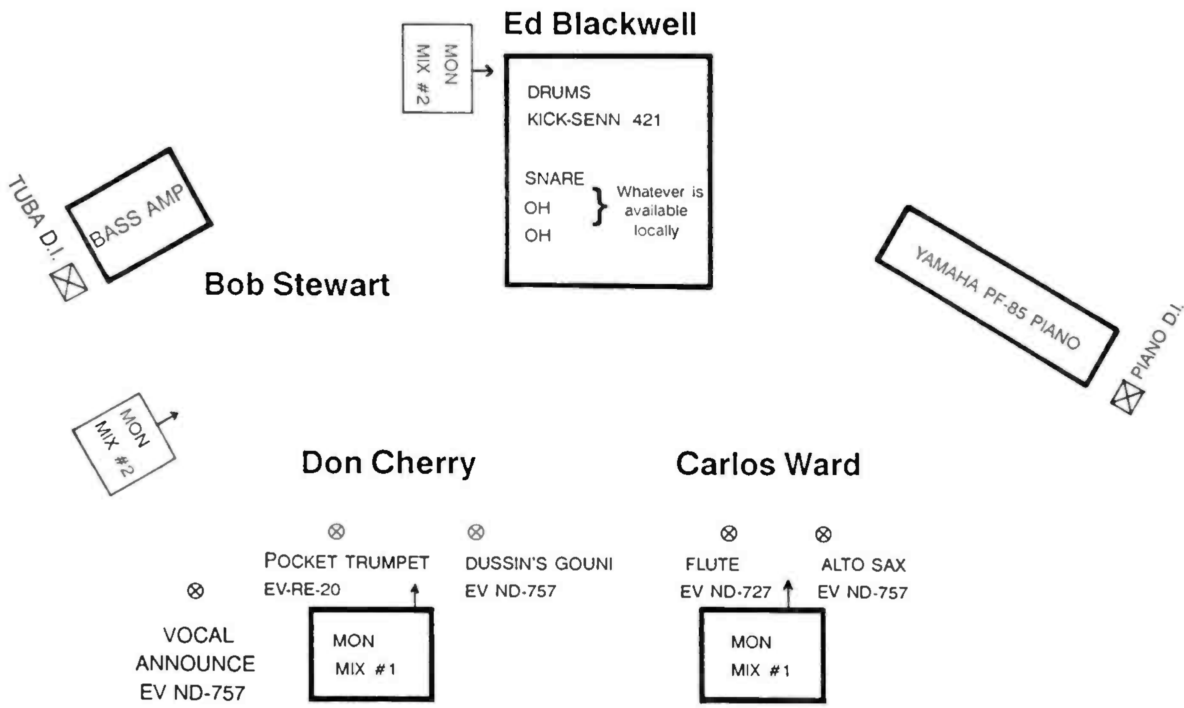


Figure 2. Floor plan and mic layout of the group.

A FAMILIAR EXTERIOR

Sunday was our first performance, and I was eager to get an early start. Pauline had the agency sound gear delivered to the theater in the morning, and I arrived there around 3 pm, a good 2 hours before the group. Located in the heart of Managua, the National Theater looked very much like the Kennedy Center in Washington, and for good reason: they were designed by the same architect. The interior was divided into several venues of differing sizes; we would perform in the 300-seat Edgar Munguia Experimental Theater. This theater was a square room, with bleachers on three sides for seating. The open side had a curtain, wing space, and ceiling bars with lights but no stage; we would set up on the floor and play to the bleachers. The acoustics were quite nice: the room was surprisingly dead, with a reverb time of around 1 second, and had a very even response. Power was available from U.S.-type grounded receptacles on the rear wall and in both dressing rooms. I measured the voltage at between 128-130 volts, and noticed the dressing room receptacles were wired out-of-phase with the "stage" receptacles.

The hardest part of setting up the sound system was figuring out what case everything was in. In deference to the small room, I elected to go without the Bose sub-woofers; I ended up using only one 802 cabinet per side, elevated on a tripod stand (*Figure 1*). I had just begun to EQ the system when the band arrived for a rehearsal/sound check. I could now establish microphone preference for each musician, and get a grip on mixing this unique group (*Figure 2*).

Don Cherry's major instrument was his pocket trumpet, which he would play both with and without a mute. His sound had a wonderful quality of warmth, so I elected to use my E-V RE-20, a large-diaphragm dynamic that really lent itself to capturing this type of sound. Its major drawback was large size and weight. Don preferred to play facing down slightly instead of facing out at the audience so I didn't need to worry about potential boom stand sag; I could use this mic on an un-extended straight mic stand. For announcements and Don's occasional vocalizing, I used an E-V ND-757 with a foam blast filter. This mic gave me the reach to pick up Don's voice even if he strayed off-mic slightly, something he would occasionally do. The doussin' gouni posed the most difficult mic'ing problem. It put out and amazing amount of low end; there were also specific overtones to pick up, and the metal scroll on top of the neck would rattle sympathetically with certain notes. Don mic'd the doussin' gouni from a hole in the side; after trying several different mics, we settled on a ND-757 with a slightly thinner blast filter. I engaged the mic's low filter to smooth out the low end, and found that by leaving the trumpet mic hot I could pick up enough of the scroll rattle to be appropriate. In the scheme of our performances, Don played the doussin' gouni solo with vocal accompaniment, so I didn't have to worry about bleed from other instruments. Don also played some piano during the show, using the Yamaha keyboard we carried with us. The PF-85 keyboard had an amplifier and two small speakers built in; it put out just enough level to be adequate for Don's needs. I used a direct box to pick the keyboard signal off the line output.

I'd previously worked with two groups that featured Carlos Ward: B.T. Express, a pop/funk group with a contemporary sound, and Abdullah Ibrahim's big band, which featured orchestrated jazz. This small sampling illustrated the wide range of Carlos' musical vocabulary. His talents as a composer were also prodigious: several of his compositions were performed nightly by the quartet. On this tour, Carlos played alto saxophone and flute; I gave him a separate mic for each. This made switching between instruments much faster, and allowed me to EQ mics differently for each instrument. I used ND-757 mics with blast filters for both; Carlos usually asked me to roll off some of the high frequencies on his alto sax. The flute mic was basically run flat, with just a bit of the lows pulled out to reduce breath noise. Boom mic stands were preferred for both, although a gooseneck was acceptable for the sax if our stand complement was limited.

TUBA IN JAZZ

Bass plays an important role in music: low notes contain the power to support a composition, and, in the jazz idiom, they can add unique rhythmic, harmonic, and melodic implications. Imagine the challenge of playing this style on a tuba: a continuous flow of bass sounds must be maintained in support of the melody and solos, and it must be done for an entire concert. The instrument is physically demanding to play, yet breathing must not interrupt the rhythm of a composition.

This was the challenge faced by Bob Stewart every night. A man of tremendous physical stamina, he was put to the test by this task. Bob believed that the "demise" of the tuba as a major component in modern rhythm sections came about as a result of increasing the volume of music. As amps and other instruments created more level, the tubist was also required to play louder. The instrument was already so physically demanding that the ante could no longer be raised; amplified basses became the answer. Bob's simple solution: become amplified himself! He used an Audio-Technica ATM-21 dynamic mic, which he suspended in the bell of his tuba via rubber straps. This mic was run into his direct box through the use of a low Z-high Z line transformer. At that point, the signal was split to feed Bob's amp and the PA system. Bob preferred an electric bass amp; we discovered the hard way that most guitar amps couldn't handle the large transients that Bob was capable of delivering. With a little EQ, he could get that amplified tuba plenty loud without stage feedback. Without having to waste energy attaining volume, Bob could conserve his strength for playing music. The sound I got was excellent; Bob usually wanted me to reduce the highs just a touch, and in some places we might reduce low end just enough to prevent the sound from becoming too boomy.

Ed Blackwell has the respect of drummers everywhere; his approach to percussion has been called the "most advanced in jazz to date." I'd first worked with Ed during his tenure with

Figure 3. Neysy Rios Auditorium, Central American University. Lois and Toney roll out the drum rug.

Old and New Dreams, where he made an indelible impression on me. Ed's encyclopedia knowledge of rhythms and sense of colors struck me as salient points in his approach. I wanted the balance between drums to be dictated by Ed, not me, so I decided to use minimal drum mic'ing. Ed played Sonor drums, with the kit based around an 18-inch bass drum, 14½-inch metal snare, 9x13-inch mounted tom, and a 14-inch floor tom. His cymbals were all Paiste, and included 14-inch hi-hats, 22-inch ride, 18-inch crash, and an 18-inch China-type. Our concerts in Nicaragua were held in such small venues that I never needed to mic the drums; I would let Ed play and balance the band to him. For larger venues, I mic'd the bass drum with my Sennheiser 421. For the rest of the set, I used the best mics we could procure locally. I would round out drum mic'ing with a snare mic and two overheads at larger venues, but most often went with just kick-overhead.

PERSONAL MONITORING

Monitor placement for our group was obvious: each musician had his own wedge. The actual distribution of monitor mixes, however, sorted itself

Figure 4. Ed Blackwell prepares for the second Managua concert.

out during our first soundcheck. Don and Carlos wanted to hear themselves on top with just a taste of Bob. With careful placement, I could not only cover them with the front two wedges, but get enough horns in the back to satisfy Bob and Ed, too. As a result, all Bob needed was tuba. This necessitated a separate mix, comprised only of tuba. Bob also wanted me to "crank it up" during the last few tunes of any performance, when he began to get tired. Ed could hear the horns fine off the front wedges, and just needed a little of Bob. I ran his wedge off Bob's mix; that was good, but a little too loud. I solved this discrepancy by turning down the amplifier channel feeding Ed's wedge. If I didn't have the luxury of dedicated amp channels/wedge, I would simply move the wedge away from Ed until we achieved the desired drop in level.

What I heard at sound check served notice that I had a hot band on my hands! Our 7:30pm concert drew an overflow crowd of 400 that packed the place, even covering the floor. The group played the room perfectly; acoustic drums and tasteful playing made good sound a snap. Ed and Bob were swinging hard, with both Don and Carlos taking off! I was reluctant to modify this gorgeous sound, but a lovely flute solo by Carlos on his own *Leto* immediately tempted me to add a tasty touch of reverb via a Yamaha SPX-90. One doesn't take liberties with sound, however, so I asked the guys about my effect during intermission. They loved it, and told me to "do what's appropriate, subtly." I used the reverb sparingly, and, when Don performed his solo "doussin' gouni blues," I faded in some delay on his voice that matched the rhythmic cadence of his song. He really loved that, and told me after the show to do it on a regular basis. The audience was dazzled, calling the group back for an encore; many stayed to chat after the show. I tore down quickly with the assistance of willing audience members, and was soon back at the hotel toasting our opening night with the guys.

MEDIA PERFORMANCE

Don and the group were kept very busy on Monday with several radio and TV interviews. My workday began a little later: at 3pm, I took the gear over to the Central American University, where we would perform at the Neysy Rios Auditorium. This proved to be a large lecture hall with the "stage" at one end of a rectangular room; the sides were louvered and open to the air (*Figure 3*). All room surfaces were hard and reflective, and the "open air" design did little to offset room reflections: reverb time was around 1-3/4 seconds, and room sound was overly bright. I located many U.S.-type AC receptacles, but the only ones with functional grounds were on the stage right wall facing. This room was even smaller than last night's; I again used only 1 Bose 802 cabinet/side with no subwoofers, and let the drums go acoustic (*Figure 4*). I was forced to roll high-end off almost everything in an attempt to deal with the room's over-enhancement. The monitors were also run substantially hotter as the group struggled with room reverb. Fortunately, the audience rescued us—we again had an overflow crowd, which helped dry up the room considerably. Capacity was purportedly 450, yet university officials pegged our primarily student audience at around 1000. Not only was there not one square inch of room inside, but people were lined up three deep outside, listening and trying to watch through the wall louvers. Things were going great until the house right PA speaker cut out halfway through the second set. I had to pick my way through masses of seated students to reach the power amps, which were hot enough to fry eggs! The system amp rack contained 1 QSC 3350 and 2 QSC 3500 amplifiers in an Anvil case just big enough to accommodate the three amps. With no space in between for air flow and no fans, it was no surprise the amps were going into thermal protection. In short order, the other working amps shut down as well. While I silently cursed whoever had designed this rack, I rapidly combined loads and switched to unused amp sides. I was forced to repeat this procedure four more times to complete the show; as one amp side overheated, another would cool off enough to use. In this sweltering room, the amps never stayed cool for long; I guess the air-conditioned environment of the National Theater had saved me from a similar fate there. I explained the problem to Toney and Pauline; they agreed that the rack should be returned to Washington for modification. In the meantime, Pauline arranged for me to borrow a small desk fan from her office to tide me over.

Tuesday featured a lunch in our honor at the Casa Grande; this fabulous home, perched high on a hill overlooking Managua, is now used as a guest house by the U.S. Embassy. Our performances were the talk of the town: Lois informed us that the embassy phone was ringing off the hook with requests for passes or tickets. I returned to the friendly confines of the Edgar Munguia Experimental Theater later that afternoon to prepare for our second concert there. Nicaragua's Radio Sandino planned to broadcast the concert live, so I arranged to meet with their engineers at the hall. Surprise, surprise! They were the same guys who had done the Wayne Toups broadcast with me. It so happened that we were all wearing our Zydecajun t-shirts, which led to a lot of good-natured joking! We decided to do what had worked so well during my last visit: they took the band's audio feed from me, using their own mics for ambience and radio announcements.

Our third consecutive capacity-and-over crowd was primed for this concert, and, since the guys were already acclimated to the theater, they were inspired to new heights. The audience was a unique cross-section of society: Sandinista officials, Nicaraguan students, internationalists, and Americans all gathered to spur the band on with raucous applause. Ideological and cultural differences were set aside in the enjoyment of exceptional music. Don improvised a "Managua blues" during his doussin' gouni solo; Bob shocked the audience with several "elephant cries" on his tuba. Carlos ripped off some fiery solos, with plenty of support from Ed's bubbling beat. I had my hands full doing two mixes simultaneously: the radio feed was radically different in balance and content from the live PA mix. I'd mic'd the drums, but only assigned them to the radio buss. This necessitated more horn level to keep up; I had to be especially alert during solos to prevent the horns from dropping out or becoming too loud. I monitored the radio feed via headphones, removing them frequently to check the live mix. Frequent changes were essential for good control. I'd asked IO (information officer) Alberto Fernandez to tape the broadcast on cassette for us, and we were able to listen back to it that night after we returned to the hotel. Don told me I'd done a good job on the mix, and everyone agreed that the effects were appropriate. I was fairly sure I had it down at this point, but it always helps the confidence to hear it from the group. db

Lab Report

SOUNDCRAFTSMEN MODEL 300x4 POWER AMPLIFIER

GENERAL INFORMATION

The Soundcraftsmen 300x4, as its name suggests, is a four-channel amplifier. No, Soundcraftsmen is not suggesting a return to so-called "quadraphonic" sound reproduction in professional applications. Rather, one of the chief advantages of this unusual configuration is its flexibility. It allows the user to select either two-channel, three-channel or four-channel operation. It makes an ideal amplifier, for example, for the large recording studio that requires very high power for monitoring. Used in the two-channel bridged mode, it will deliver 600 watts per channel into 8 ohm loads. Alternatively, you can bi-amp your monitors (using the amp in its four-channel mode) to obtain 210 watts per channel at 8 ohms. Still another possibility is tri-amping, using two of the Model 300x4 amps, each in the three-channel mode, to provide 600 watts per channel for the woofers and 210 watts per channel for the high frequency drivers.

The 300x4 occupies the equivalent of three rack spaces in height. It has two completely independent power supplies and two separate power transformers. The 300x4 is completely protected against short circuits and input overloads. Thermal protection is provided by multi-sensor phase control regulation as well as two multi-speed cooling fans.

The basic circuitry of the 300x4 is similar to previous models offered by Soundcraftsmen. The company is convinced that power MOSFETs are not only more reliable than other output devices (they are self-limiting as far as dissipation is concerned) but also provide what some have described as "tube-like" sound quality that many professional sound equipment users consider to be superior to the "transistorized" sound of bipolar transistors.

In many respects, the circuitry of the 300x4 is almost identical to what you would have if you installed a pair of Soundcraftsmen Model PM860 amplifiers in a single rack-mount panel (their Model PCX-2 rack-mount front panel), except that the designers of this four-channel/three-channel/two-channel amplifier have beefed up the power supply and transformer capabilities so that the amplifier can truly be considered a low-impedance, high-current, high-wattage unit.

Figure 1. Frequency response.

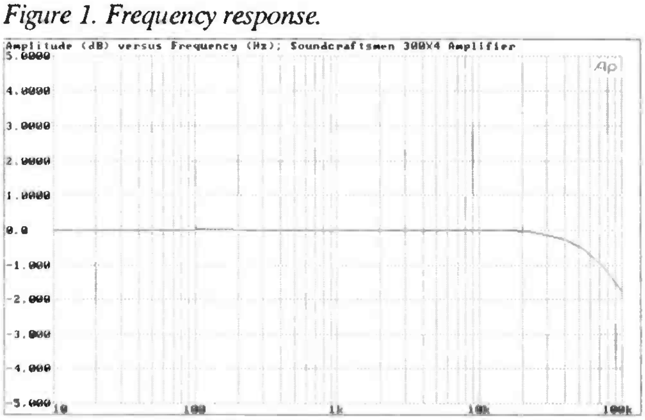
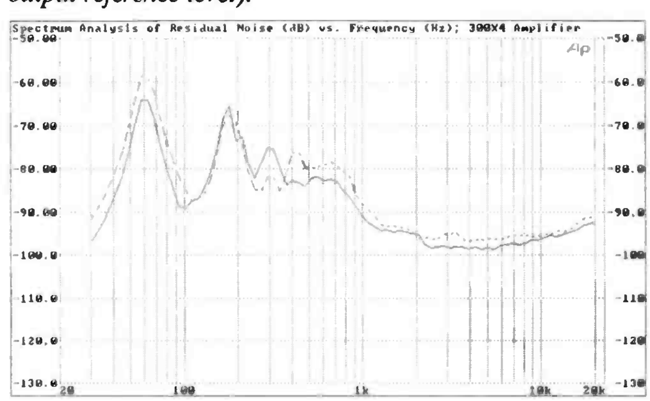


Figure 2. Spectrum analysis of residual noise (1-watt output reference level).



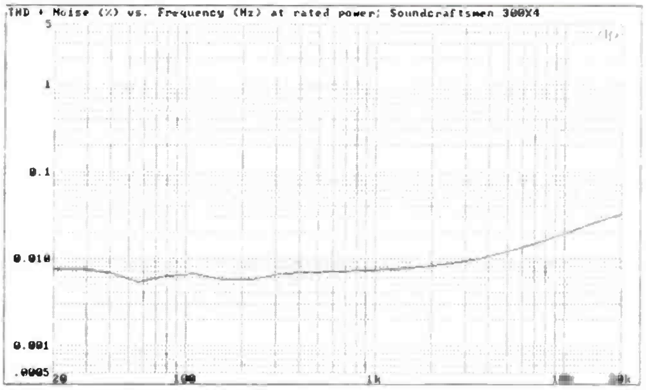


Figure 3A. Harmonic distortion plus noise versus frequency, at rated power (210W/channel, 8 ohm loads).

The amplifier that we tested was, in some ways, a prototype model with some minor changes in circuit board layout and wire routing still to be finalized in production. We were assured by Soundcraftsmen, however, that all major components as well as the high current, low-impedance power transformers used in our sample are the same as those that will be used in final production. Frankly, we don't know why Soundcraftsmen even bothered to mention the prototype aspects of the sample; to us, the "innards" of this amplifier looked a lot better than those of some models presented to

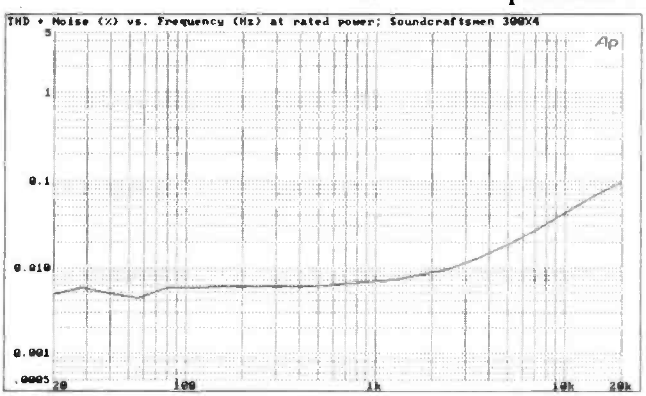
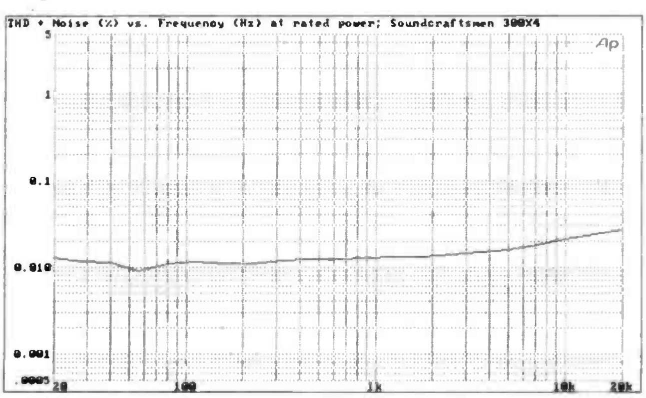


Figure 3B. Harmonic distortion plus noise versus frequency, at rated power (300W/channel, 4 ohm loads).

us as actual "production models."

Figure 3C. Harmonic distortion plus noise versus frequency, at rated power (600W/channel, 8 ohm loads bridged mode).



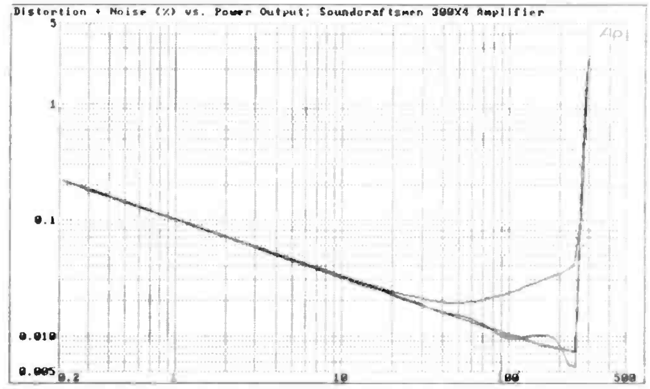
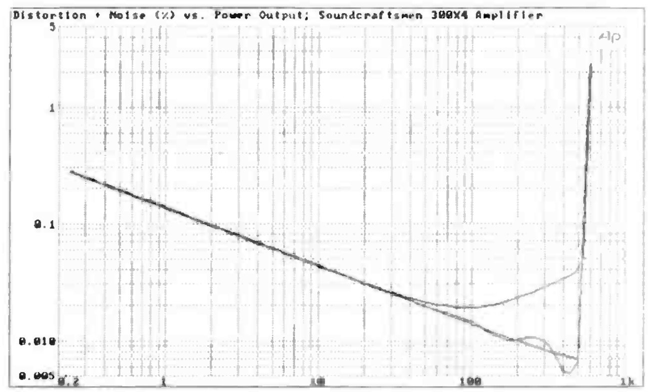


Figure 4A. Harmonic distortion plus noise versus power output per channel (8-ohm loads). 20 Hz and 1 kHz, lower curves; 20 kHz, upper curve.

FRONT AND REAR PANEL LAYOUT

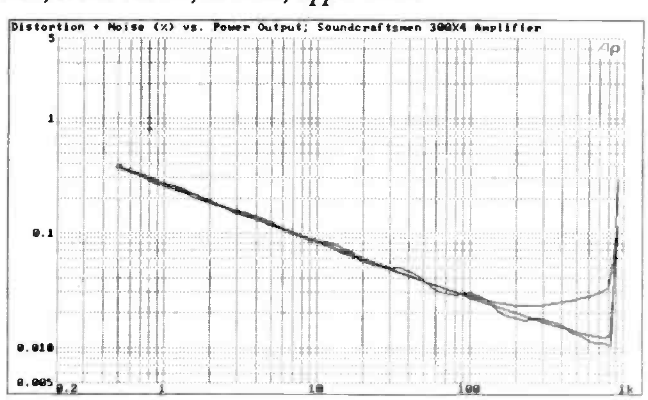
The only control found on the front panel is the power switch. When activated, there is a momentary delay before full operation of the amplifier is available. Banks of green LED indicators (three on each side of the amplifier's front panel) tell the user which mode is being used. Two of the LEDs on each side illuminate when the amplifier is operated in the four-channel mode while the third LED of each bank



Fiigure 4B. Harmonic distortion plus noise versus power output per channel (4-ohm loads). 20 Hz and 1 kHz, lower curves; 20 kHz, upper curve.

illuminates when the amplifiers have been switched to the bridged mode. Additional red LEDs illuminate when amplifier clipping or overdriving takes place. Finally, there are also two circuit-breaker reset buttons that are used to restore amplifier service in the event that these breakers open

Figure 4C. Harmonic distortion plus noise versus power output per channel (8-ohm loads, bridged). 20 Hz and 1 kHz, lower curves; 20 kHz, upper curve.



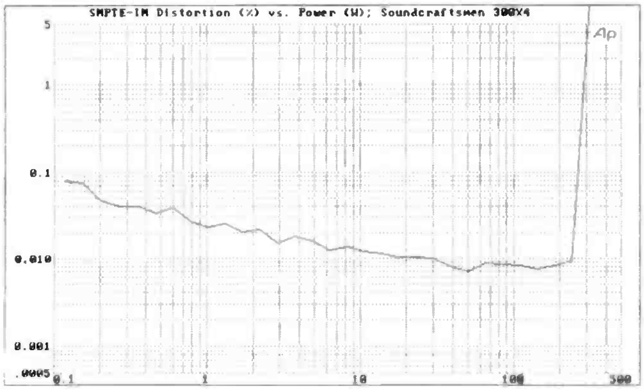


Figure 5A. SMPTE-IM distortion versus power output, 8-ohm loads.

for whatever reason. During the entire course of our lab testing, it was never necessary to use these reset buttons, a fact that attests to the ruggedness and reliability of the amplifiers. We probably subjected them to more stress than they are likely to get under normal operating conditions when fed with music or speech signals.

The rear panel of the 300x4 is equipped with four sets of color coded five-way speaker binding post terminals. When the amplifiers are bridged (for two channel, stereo operation), connection of load is made from red terminal to red terminal of the same amplifier pair. Otherwise, connection of loads is made from red to black terminals. Slide switches are provided for selecting bridged or 4-channel mode, so that no internal wiring changes need to be made when going from one mode of operation to the other.

One surprising aspect of the design was the absence of balanced inputs commonly found on amplifiers intended for professional use. The only inputs available are unbalanced type, via ¼-inch standard phone plugs. That's certainly a step above those consumer type "phono tip" jacks, but short of the more rugged 3-pin XLR connectors which might have been used even if only unbalanced inputs were provided. There are no level controls on the rear panel, and input sensitivity is therefore fixed at around 1.3 volts for rated output, when in the unbridged, four-channel mode.

LAB MEASUREMENTS

The 300x4 amplifier exhibited flat frequency response from below 10 Hz to 20 kHz, within 0.1 dB. The graph of *Figure 1* illustrates that response extended beyond the audio spectrum and was down by only −1.7 dB at 100 kHz. Response was virtually identical for all four available channels or, in stereo bridged mode, for both bridged channels.

Referred to 1 watt output, A-weighted signal-to-noise ratio measured 77 dB. Translated to a rated output of 210 watts, the S/N ratio would be just over 100 dB. This falls a bit short of the 105 dB claimed by Soundcraftsmen for the similar PM860 amplifier, whose specifications are supposed to be similar to those of the 300x4. The reason for this may well have to do with the prototype nature of our sample. As can be seen in *Figure 2* (a spectrum analysis of noise versus frequency of this amplifier), substantial peaks of noise show up at the power-line frequency (60 Hz) and at its third harmonic. No doubt the final wiring layout, of which Soundcraftsmen told us, will bring the S/N ratio to well within the rated spec.

Figure 3A shows how total harmonic distortion plus noise varied as a function of frequency, with input regulated so as to produce rated output (210 watts per channel) into 8 ohm loads. Under these conditions, THD plus noise at 1 kHz measured only 0.0072 percent at 1 kHz, virtually the same value at 20 Hz and a somewhat higher 0.031 percent at 20 kHz. Even at that upper frequency, THD plus noise was far below Soundcraftsmen's rated value of 0.05 percent. The same sort of measurement was repeated for 4 ohm loads, with input regulated to produce a constant 300 watts of output per channel. Results are shown in *Figure 3B*. At mid and low frequencies, THD plus noise for these loads remained extremely low, while at 20 kHz, we measured a THD plus noise figure of 0.095 percent. Finally, the same measurement was repeated for the bridge mode, using 8 ohm loads and regulating the input so as to produce a constant 600 watts per channel. Under these conditions, THD plus noise remained slightly higher than 0.01 percent at low and mid frequencies, rising slightly to 0.027 percent at 20 kHz, as shown in *Figure 3C*.

Figures 4A, *4B* and *4C* present plots of THD plus noise as a function of power output for the three conditions described. The apparent higher percentages of THD + noise at low output levels are not, in fact, harmonic distortion but rather noise contributions referred to those lower levels.

Figures 5A, *5B* and *5C* show how SMPTE-IM distortion varied as a function of power output levels for 8 ohm, 4 ohm and bridged (8 ohm) operation. In this case, the percentages remained well under 0.05 percent at rated power levels (210 watts for the 8 ohm/4 channel mode, 300 watts per channel for the 4 ohm/4 channel mode and 600 watts per channel for the bridged, 8-ohm stereo mode.)

Figure 5B. SMPTE-IM distortion versus power output, 4-ohm loads.

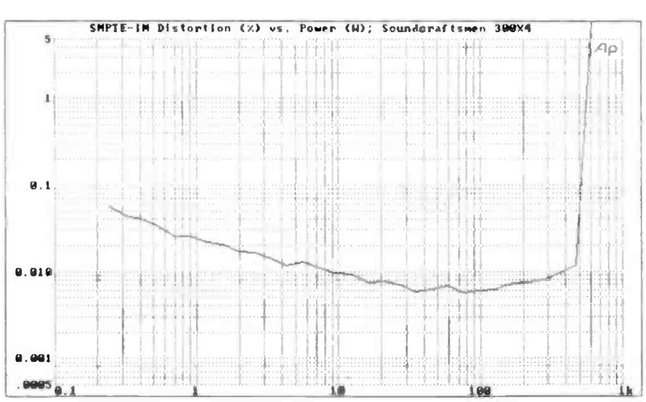
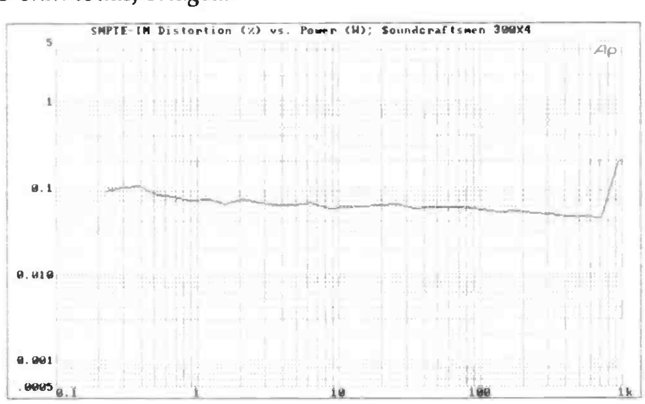


Figure 5C. SMPTE-IM distortion versus power output, 8-ohm loads, bridged.



Input sensitivity of the amplifier, referred to 1 watt output, measured 87 mV in the bridged mode. Damping factor, referred to 8 ohm loads and using a 50 Hz test signal, measured in excess of 200. Dynamic headroom, using 8 ohm loads, measured just over 1.0 dB.

COMMENTS

As has been true in tests of other Soundcraftsmen power amplifiers, their chief attributes have been excellent reliability and superb sound quality. Many an amplifier that we've tested over the years sounds good when driven to moderate levels with music signals but doesn't stand up in the field where it is often pushed to its limits. Our bench tests probably subjected this Soundcraftsmen amplifier to conditions that it is not likely to encounter even in the most demanding sound reinforcement applications. Yet, the amp never faltered, never shut down and, amazingly, didn't get all that hot either.

As for sound quality, we completely agree with Soundcraftsmen's engineers as far as their choice of MOSFET output devices is concerned. For that matter, there are a lot of amplifiers out there that employ far-out tricks to bring measured distortion levels down to "triple zero" numbers of distortion following the decimal point, but when you hook them up to real-world speaker loads their sonic quality falls short of matching the bench test numbers. Not so with the Soundcraftsmen 300x4. To be sure, the amplifier's THD figures are not the lowest on record, but if there is anyone out there who can tell the difference between 0.05 percent THD and 0.005 percent THD we have yet to encounter such skills! By taking a straightforward approach to amplifier design, Soundcraftsmen has consistently been able to deliver amplifiers that perform reliably and are well suited for the kind of use to which they are likely to be put in studio monitoring situations as well as in sound reinforcement applications. Best of all, the 300x4, like other amplifiers from this company, is fairly—make that more than fairly—priced compared with the competition. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db Measured
Power Output 8-ohm/4 Ch.	210 watts/ch	230 watts/ch
4-ohm/4 Ch.	300 watts/ch	420 watts/ch
8-ohm Bridged	600 watts/ch	720 watts/ch
Rated THD(8 ohms)	0.05%	0.032%Input
Sensitivity	1.2 Volts	1.3 Volts
Damping Factor	More than 200	Confirmed
Signal-to-Noise	105 dB	100 dB (See text)
IM Distortion	0.05%	0.009% (8 ohms)
Dynamic Headroom	N/A	1.02 dB
Dimensions (WxHxD)	19x5-1/4x12 inches	Confirmed
Weight	58 Lbs.	Confirmed
Price: $1299.00		

THE ELECTRONIC COTTAGE

Mike Mandel: The Songwriter's Dreammaker

• Everyone who would succeed in life must carve out a niche—a space where one's skills can be seen as unique in the marketplace. This reality is doubly true in the highly competitive world of the professional musician. Electronic cottage facilities have provided some musicians with a new, and virtually unlimited arena for their talents, where their hard-earned *chops*—the product of numerous grueling years on the road—can be profitably utilized without ever having to leave their home. One such musician is Mike Mandel.

My recent encounter with Mandel's production skills started in the living room of a friend. A few weeks earlier she had played me some rough demos from a songwriter she knew. The songs were well-crafted but needed some serious arrangements and production in order to be professionally acceptable by today's sophisticated standards. She explained how this songwriter had struggled for years to get a publishing deal, all to no avail until recently, when on the strength of some new demos he had paid someone by the name of Mike Mandel to produce for him, he hit paydirt. Today she wanted me to hear these new demos.

PLEASANT SURPRISE

She was excited that her friend had finally signed a publishing deal, but I was curious as to what kind of production had made the difference. By telling me about a producer from New York City who transforms songs into commercially viable entities for a very reasonable fee, she had piqued my interest. Cynically, I expected a quick-fix, bang-em-up MIDI production, but when I finally heard Mike Mandel's treatment of these songs I was quite impressed.

What I heard was something beyond competent production. It was subtle, nuanced, the evidence of not only a good programmer, but a master musician and arranger as well. The name Mike Mandel sounded familiar, but I couldn't place it. So I decided to pay a visit to this songwriter's dreammaker.

Mandel's electronic cottage is located on the 20th floor of a beautiful apartment complex on Manhattan's west side. Perched high above any neighboring buildings, it offers a clear view out to the Hudson River—a feature which is undoubtedly inspirational to his clients. Mike Mandel however, is not at all concerned about the view from his studio. His source of inspiration comes from other places.

He has been blind since early childhood.

Arriving at the studio, I was greeted by Mandel and his guide dog, Clyde. We spent a very pleasant two hours talking about production techniques, philosophies of music and the special requirements of outfitting a studio for an unsighted operator. Additionally, I found out a little bit more about Mike Mandel and why his name had such a familiar ring.

As it turns out I had seen Mike Mandel before, perhaps several times. I found out that he had done a lengthy stint with radical jazz guitarist Larry Coryell (and the Eleventh House). I had seen the group on several occasions, and was extremely impressed by this blind keyboard virtuoso. By sheer serendipity, it turns out that we lived on the same street in Boston during the early seventies, while we both studied at Berklee College of Music.

Figure 1. Mike Mandel composing at the piano keyboard.

Evidently, Mandel took his training much more seriously than I did. After Berklee he spent three years at the New England Conservatory, on to the Boston Conservatory, followed by private studies with noted arranger, Don Sebesky.

MUSICAL BACKGROUND

In addition to all the academic plaudits, Mandel has been involved in a pretty diverse array of musical projects. He has solo albums on the Vanguard label, has recorded with artists as diverse as Deodato and Glen Campbell, done scads of commercials for some rather upscale clients, provided source music for all your favorite soap operas, been a product specialist for Casio synthesizers, and toured with rock bassist Jack Bruce and with vocalist Roy Orbison.

With that breadth of experience under his belt, what could he be up to next? Mandel feels that the diversity in his musical background really comes together best in the context of his electronic cottage. It is here that he puts it all to work, producing songs for other writers with the authentic flair of one who has actually "lived" that musical style.

Since coming off the road, Mandel has built a thriving business out of his EC. Perennially servicing several clients at the same time, his only fear is not being able to turn around a production in a timely fashion—something which only rarely occurs. But even with his busy schedule Mandel seems unusually sensitive—almost altruistic—towards his clients' needs. Aware that struggling songwriters are not a wealthy lot, he maintains a very affordable rate structure, so that writers will be encouraged to demo their songs several times a year. By keeping to a set fee, with no hidden extras, Mandel's clients know what to expect to pay for a final product and can more easily project their long range budget.

COUNSELING THE CLIENT

A big feature of Mandel's service is the first "sit-down" meeting with the client—a service for which there is never any charge. The client will show him a rough version of the song and Mandel will fire up his drum machine and a few synthesizers and give the client his impression of which direction the song ought to go. This includes exploring rhythms, tempos, sound colors and textures and that indescribable substance called "feel." This is all done in real-time, essentially a live performance for the client. If the client is pleased with the direction, then this initial meeting ends and Mandel proceeds with the production privately. Treating each song as a unique puzzle to be solved, his production style rests on a very mystical foundation. Says Mandel:

"I believe in building moods and atmospheres for a song. Each song, besides having a distinct groove, has a distinct mood and atmosphere. I think that's what distinguishes one song from another. Not just it's point of view about the lyrics, but the atmosphere it establishes—the way it touches a human being."

Now all alone in his electronic cottage, Mandel begins to create those "moods and atmospheres" utilizing some of the classic tools of the trade. His keyboard array is modest by some standards, but evidences judicious cost-effective buying. Essentially, all the bases are covered.

STEP-BY-STEP

The Yamaha DX7-II, with its distinctively clear FM sounds serves also as a master keyboard for Mandel. The Ensoniq ESQ-M adds the unique strengths of wavetable synthesis, while the Roland D-550 fulfills the need for sampled sounds. Yamaha's TX 81-Z provides some nice extra FM sounds and flexible voice allocation all in a box, and the ubiquitous Roland D-50 contributes its characteristic breathiness. And since no modern digital gizmo can ever replace the nice, fat sounds and real-time programmability of analog synthesis, Mandel has wisely included a Sequential Circuits Prophet-5 and an Oberheim Xpander to his arsenal.

While the production method might vary for different musical styles, it usually starts with Mandel establishing a groove on his Korg DDD-1 drum machine. (He also uses the Alesis HR-16 as a drum expander to add versatility to the sound.) A bass-line is then established and recorded into his sequencer, utilizing the rich timbres of the Prophet-5 and/or the Xpander as sound sources.

Once the groove is happening, Mandel starts fleshing out the sound textures which form the foundation for the "moods and atmospheres" he is seeking. A typical step in this process would be to lay down a pad (a diffuse wash of sound which will serve as a sort of musical "glue" to hold the produc-

Figure 2. Racked keyboards partially obscure the high-rise view beyond.

Figure 3. Tape recorders and mixers all arranged as a desk-shelf convenience for Mandel.

tion together). Typically for Mandel, this might involve a combination of sounds—perhaps one of the notable electric piano sounds from DX7-II combined with complementary ethereal sounds from the ESQ-M or the D-550 —hard panned in opposite directions.

FROM THE SEQUENCER

From this basis, various colors, rhythms and counter-melodies are added—all stored in the sequencer for later manipulation. When the production is ready for vocals, the client, vocalist and possibly a live instrumentalist such as a guitar player, are brought in for a recording session. The night before the session, Mandel, who prefers doing his initial drum programming directly on the drum machine, executes a MIDI data dump (of the compiled drum patterns) into his software sequencer.

The sequencer is then locked to his multi-track machine (an Otari MX-50/50-8 track) by laying down a sync-tone. The sync-tone Mandel uses is not the standard FSK tape sync which is output by all sequencers and drum machines. The standard variety locks up well enough, but since it contains no location information (other than start/stop), it is not possible to go over a passage of the song without starting all over from the beginning. In recent months several companies have begun manufacturing interface boxes which directly translate the MIDI clock in a sequencer into a new variety of FSK which contains continuous MIDI song pointer references imbedded in the code, thereby enabling the user not only to synchronize to tape, but to start and stop sequences along any point on tape. While SMPTE time code has long offered the same feature (and many more as well), this new breed of FSK generator/readers does everything a songwriter or music producer could want it to do. Mandel uses a box by JL Cooper called the PPS, sometimes referred to as "poor man's SMPTE" (it lists for only $199). Mandel has nothing but praise for this highly reliable product. "If I go video hook-up, I'll go SMPTE. But for this (songwriter demos), I don't need it. PPS works like a dream."

Following the live session, Mandel will take a day or so to examine the tracks at his leisure. He may choose to add sweeteners at this time or perhaps, since all synthesized tracks remain as "virtual" tracks, he may experiment with various combinations of synthesizer patches, changing the programs to provide the best possible complement to the vocals. The sounds and textures remain flexible right up until the final mixdown. Only live tracks and drums are printed to tape. Strictly speaking, the drums don't have to be printed either, but since he likes to process each element of the drum kit separately, it is usually preferable to do so while tracking instead of tying up signal processors during the mix.

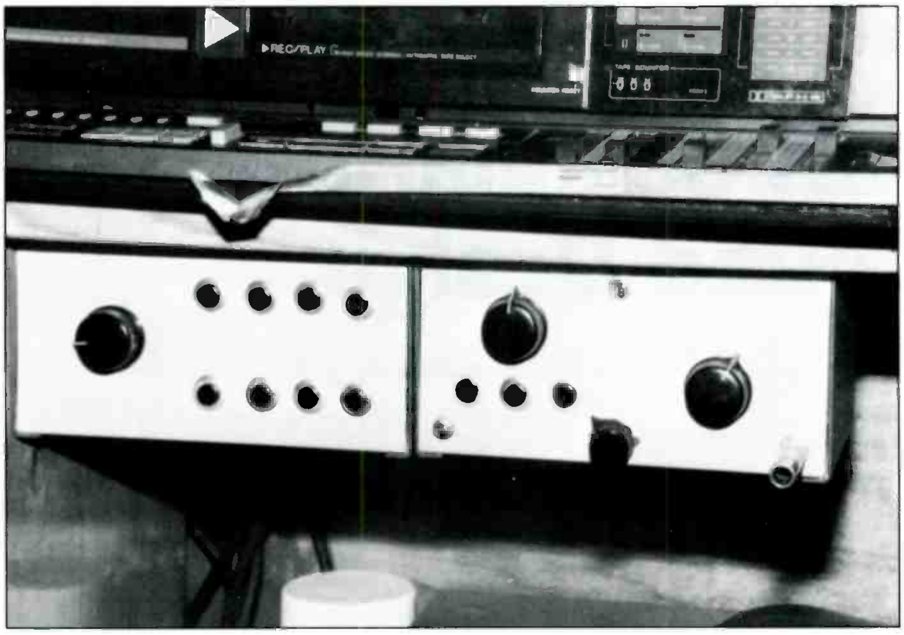
Mandel usually mixes on Yamaha NS-10M speakers, but also has a weird-looking custom made pair of speakers for a hypey low-end/zizzy high-end type reference. (They are 5-way speakers containing 12-inch, 8-inch and 5-inch speakers, plus a tweeter and super-tweeter.) Mandel reserves those speakers for clients who are dance music aficionados and want to hear a sound that will, as he puts it, "rattle your teeth." There are however, limits to how much teeth rattling can go on in an urban apartment building, so these speakers cannot be monitored for extensive periods of time. Finally, as a matter of course, Mandel routinely processes all his mixes through the Barcus Berry BBE Maxi, another in the proliferating number of aural exciters on the market. He likes the subtle harmonic transformation that goes on with the BBE and feels that this gives his mixes a little extra punch.

THE TALKING COMPUTER

I was interested in finding out how an unsighted person interacted with his computer (and for that matter, how he interacted with all the audio equipment in his studio). I asked Mandel to give me a demo of how his system worked.

The first obstacle a blind computer operator faces is learning the software. Software generally comes with a printed manual—sometimes the size of an urban telephone book—to help you learn to operate the program. Blind people do not have this option, so they have to be resourceful. They have several options before them: 1) they could pay a sighted person to read the manual onto a tape for them 2) they could see if any public or private organization for the blind would have a tape of the manual on file for them to purchase, or 3) they could check with the manufacturer to ascertain if an ASCII file of the manual is available.

Figure 4. This specially-designed box permits Mandel to hear his audio switching.



For Mandel, the third option is the most preferable. Using a Leading Edge computer (IBM compatible) with a 20 MEG hard drive, a screen reading program and speech synthesizer card (with appropriate software to drive it), Mandel is able to "listen" to the ASCII version of the manual. This same voice card/screen reader combo enables him to negotiate all the software he needs to produce music.

As a software sequencer Mandel uses both Magnetic Music's Texture 3.1 and Voyetra's Sequencer Plus Mark III. For tweaking his synthesizer patches, he uses Voyetra's Sideman voice editor for the TX81-Z, D-50 and DX-7. Voyetra scores heavily with Mandel as he also utilizes their Patchmaster voice librarian.

I sat back and watched as Mandel punched up some tracks he had been sequencing using the Texture program. Calling up a page of the program he wanted to work with, he directed the cursor—one increment down at a time—until it rested on the line he wanted to read. It contained some information on track assignments, instrument names, controller information and some other things I'm not quite sure of. When he instructed it to read, it would read the entire line from left to right (unless he instructed it to do otherwise). Faithful servant that it was, it read everything—not just words, but all alpha numeric info including punctuation. Sounding a bit like a native-born Rumanian android, it spit out a phrase that sounded something like, "RIM-SLASH-1A- SLASH -3-DOT-DOT-DOT-3-DASH-1-9-5."

Mandel said it would sound a whole lot more natural if he sped it up and shut-off the punctuation command. I listened back, but it was still all Greek to me (or was that Rumanian). Nevertheless, he assured me that once one got acclimated to the sound and learned the layout of the program, it would be easy enough to interact with. Apparently, he has become very proficient at using it.

In fact, Mike Mandel interacted more quickly with his computer than a sighted person could, because he did not need to stop and divert his gaze in order to take in visual information from the screen.

AUDITORY PPM

Monitoring recording levels is almost impossible without proper metering. Thanks to an organization called Science For The Blind a special unit was designed for Mandel that translates peak metering into audible "squawks." It is calibrated internally to respond to either 0, +4 or +8 peaks. From the front panel, Mandel can switch any channel into any meter to check the levels. Depending on the nature of the sound, he may monitor for a higher or lower level. In any case, when the threshold has been exceeded, a very noticeable squawk can be heard. The sound however, is emitted through a separate monitor system so that the musical program will not be disturbed.

Certain other conveniences are built in. Like braille markers (an embossed stick-on tape) to label channels on the mixing console or to point out pre-set optimum levels for invariable things like sync tones. But frankly, Mandel says, most things must simply be committed to memory, because labels simply do not fit on the narrow surfaces of most studio gear.

Another device Mandel showed me is not strictly speaking a piece of studio gear, but it helps him organize his life and his studio schedule in a very efficient way. (Perhaps all studio owners should buy one of these gadgets!) Mandel was excited about his new special purpose lap-top computer—something called Braille and Speak.

He explains: "Braille and Speak is a 200K microprocessor that uses a 7 key Braille format entry and has speech output. So I can carry it around in my jacket pocket, I can take notes, appointments, phone numbers. It's really neat. It's only been around about a year and it's revolutionized the way a lot of us operate. We used to carry around either slates and styluses and lots of paper, or machines that were kind of clacky. Not only was entry a drag, but so was retrieving the information. Nobody wants to carry around 300 Braille pages so that they'll have their schedules and their notebooks and their seminar notes to study and a phone book. This is really great. I'm really happy with it!"

It was clear from my visit with Mike Mandel that the microprocessor revolution has made a great impact on his life. Living and working in the serene creative environment of the electronic cottage has enabled him to utilize his God-given talents quite profitably, and in so doing, make many a songwriters dream come true. db

BRUCE BARTLETT

AES Seminar on Live Theater Sound Design

At the November 1988 AES Convention, a group of experts presented a seminar on live theater sound design. If you have questions about this topic, chances are you'll find some good answers here.

The Chairman was Shelley Herman of Accutronics Sound Company. The panelists were these:

- Jonathon Deans has worked in theater sound for 16 years, and in the last nine years designed sound for over 20 productions of music and plays. He's worked mostly in England, but has done some international productions. Mr. Deans was a sound engineer for the Royal Opera House Covent Garden and Autograph Sound of London.
- Bill Hennig has 25 years experience in technical media, and has done live sound for California Jam 1, three tours and a double live album for Frank Zappa and the Mothers of Invention, the Earth Wind & Fire world tour 1975 and 1980, Seasons at the Greek Theater, Tommy at the Aquarius Theater, Los Angeles Music Center, Almonson Theater, and Long Beach Civic Light Opera: resident designer 1985 to present. He is a member of IATSE for the past 23 years.
- Al Cinescal is owner, sound designer, and chief executive officer of A-1 Audio in Hollywood, a major touring sound company handling such acts as Charo, Ann Margaret, and Frank Sinatra.
- Abe Jacob is the "father of sound design in theater." He brought a hybridization of techniques of concert sound to the musical stage. Jacob worked on such award-winning musicals as Hair, Jesus Christ Superstar, Pippin, and Les Miserables. He has done concert sound for Jimi Hendrix, The Mamas and the Papas, and Peter Paul & Mary. Jacob is the executive secretary and a founding member of the Theatrical Sound Designers Association.
- John Gottlieb is the resident sound designer at the Los Angeles Theater Center, Pasadena Playhouse, and the Taper.

The panelists' comments have been condensed and paraphrased for this article.

What Are The Job Requirements Of A Sound Designer?

The function is to project and enhance the performance on stage and deliver it to the audience with the same impact that is happening on the stage, throughout the auditorium. The designer must be ready to make compromises with artists, directors, and producers.

Have hands-on experience with equipment before you do design work. Know your tools.

Try to understand the goals of the production people, and determine what type of equipment will be needed to enhance the performance. Try to stay up with the state of the art but keep costs under control.

Design a sound system which reproduces in every part of the theater the sound as it appears at the microphones. Create special effects that enhance the mood and character of the play or musical. Ideally, the sound should be as good as that of a home stereo system.

How Did You Contact Prospective Customers When You Were First Starting?

I walked in the front door of a few theaters at the right time when they needed somebody. I convinced them that I knew what I was doing and submitted a bid.

How Do You Determine Your Fee?

The fee is based on theater size and budget—normally $2000 to $3000 for a 700-to-1000-seat show, with about one month pre-production.

What Is Your Procedure After Being Handed The Script?

Very early on, become a part of the production team by talking with the director and set designer. You integrate the speakers and microphones with the set. Fight for what you want; be aggressive about speaker locations. Advise the producer about what is required for the show and what it will cost.

How Long Do You Follow Through With A Show?

At least all the way through press night. Normally you stay until the conclusion of the show's run. Check that the operator is following your design and ask about problems.

What Are Your Considerations For Shows That Go On The Road?

We make our own cables, put connectors on everything, and carry our own power distribution. We set up in our own shop first, and do a dry run with all the equipment.

It's important to make the speaker system flexible enough to interface with any environment. Carry different types of speaker systems for different theaters.

Rely on your own equipment—assume that none of the theater's equipment works. However, many larger venues now have well-designed sound-reinforcement systems.

Do You Specify Mic Brands, Speaker Placement, Etc., Or Just Set Overall Goals And Let The Tech People In The Tour Company Determine The Hardware And Placement?

Specify every detail. It's your name on the line. However, you may have to make some changes or compromises.

When You Specify, Do You Include The Equipment Of The Tour Company And The Theater?

You can't count on the theater's list of available equipment, some of which may not work or may not exist. Whenever possible, back yourself up by specifying new equipment.

How Do You Handle One-Time Events?

Take all your equipment, including lots of spares. Keep the system simple, so that when you set up, you have time to listen and experiment with speaker locations.

How Do You Decide What Recorded Music To Use With The Drama?

Collaborate with the director. Ideally, you can take suggestions from each other, but sometimes you just follow the director's ideas.

Do You Prefer Sound Effects To Be Live Or Recorded?

I prefer recorded because a tape punch-up is more reliable. For more realism, it helps to put a speaker where a person performing the effect would have been. Stick to either live or recorded effects, because switching back and forth can be distracting.

Live gunshots and some door slams are much more effective live. A live musician can be amplified over the main system.

Do You Record Your Own Effects Or Use Libraries?

I borrow from someone else's large, expensive library. It often helps to modify or layer the stock sounds.

For Sound Effects, Do You Use Open-reel, Cassette, Broadcast Cartridges, Or A Digital Storage Medium?

It depends on the show. For a show with many cues, you can use two or three open-reel machines and a tape operator. A simple production can use carts. The standard is the NAB cartridge recorded by a production house. It's easy to use, easy to cue up correctly, and has mechanically quiet operation.

I've used a sampler for a ping-pong game.

Do You Play Sound Effects Over The Main House Speakers Or Over Strategically Located Speakers?

I put a speaker where the sound is supposed to be coming from, and play the effect through that speaker. It's more realistic and less distracting.

Often the on-stage volume of the effect is too loud for the actors, so I try to aim the speaker at the audience and away from the actors.

If the effect is the sound of the actor's thoughts, I play it through the house speakers.

What Are Your Comments On Wireless Microphones?

These days, wireless mics are often a necessity. That's because the audience has come to expect better, more intelligible sound—like a home stereo system. Also, the actors often must compete with loud background music. In a 3000-seat house, unless you completely isolate the orchestra, you must use wireless body mics—not area mics. Musicals usually require body mics; plays do not always require body mics.

The goal is to have unnoticeable sound reinforcement. So if the lead character has a wireless mic, all the other main actors will need one too, to avoid an obvious change in sound quality.

You must train actors in wireless mic technique. Make sure they project (talk loudly), and have them try to keep their volume relatively constant. Don't let them swap mics with each other! Tell them to come to you with sound problems.

Ninety percent of the mic problems are mechanical—damage to the microphone element or connector. Affix the mic in a harness or strap to limit its movements. Work with the actor in rehearsal to work out unusual mic placements.

In the Broadway theater, the majority of the wireless units are Sennheiser; in the London theater, Micron seems to be the main choice; in Los Angeles, Vega is a big name.

What Problems Arise When Two Actors With Wireless Lavalier Mics Are Close To Each Other?

You get phase cancellations, but this can be prevented by using only one mic to pick up both people. Follow the blocking; turn mics up and down as needed.

What Are Some Problems With Mic'ing The Pit Orchestra?

The musicians move their mics away from where you placed them. It helps to clip a gooseneck mic to the musician's chair or music stand. A useful product would be a microphone stand with a built-in music stand. You can clip mics to violins; just be sure that the musicians don't move the microphone. Have a second engineer on stage check the mic placements before each show.

What "Area" Microphones Do You Like To Use?

I don't use omnidirectional microphones, because they can't isolate the stage from ambience. I like AKG preamps and mic capsules. I have used AKG foot mics with CK-3 capsules. The Crown PCC-160 works great for picking up tap dancing. It has a wide field of pickup, so sometimes you can use fewer of them. You might want some mics tuned for taps, others for chorus or dialog. I like Sennheiser 421s for reeds. Also I like clip-on mics for strings, and Shure SM-57 and SM-58 for general-purpose, wide-range mics.

I like the PCCs for general ambience and vocal pickup. I like their pattern. Because they're physically flat, they tend to hide better.

Where And How Do You Use Boundary Microphones?

I use Crown PCC-160 mics across the front of the stage. They are less than an inch high so you can hide them. Because of its supercardioid pattern which rejects ambient noise, the PCC-160 has succeeded the PZM. I might use PZMs for ambience mics above an orchestra.

If you have a chorus upstage, you might use a shotgun to pick them up locally because you can't have a mic in the middle of the floor.

I've seen plexiglass boundaries put around a PZM, but it's like looking through a fence.

It seems that the Crown PCC-160 is the generally accepted microphone of sound designers for front-of-stage location for area pickup. They seem to sound best with minimal EQ. You get adequate gain if the rest of the system is equalized properly. They have a low profile and are quite rugged. So for those practical applications they seem to be the ones that are being used.

You have to learn the show, have specific mics up at specific times, make them conform to the blocking. Have only one or two mics up at a time. As you turn up more mics, you add ambience and it washes out the sound.

How Do You Use Shotgun Mics?

I hang three or four AKG CK-8 or CK-9 units on a scenery pipe upstage, or in the wings. This provides useable gain.

What Brands Of Musicians' Microphones Do You Like?

My favorite handheld dynamic is the Sennheiser 431. The performers seem to like it. The AKG 535 seems to be a great mic.

For the orchestra, I like Neumann, AKG, Shure, Sennheiser—whatever's best for the job. If the orchestra is in a separate room, there's more freedom to use mic techniques like those in a recording studio.

When You Equalize The House, What Sort Of Curve Do You Prefer?

I used a C-weighted curve for musicals. I like the Source Independent Measurement (SIM) system by Meyer Sound Labs.

The general procedure is to play pink noise through the system, run around with an Ivie ⅓-octave analyzer, and set a desired curve. Then play a CD that you're familiar with and re-equalize. You still need the pink-noise curve for a reference, but there are phase cancellations and standing waves that don't come out in the curve adjustment—you have to use your own judgement. You might use one type of equalization for foot mics, another for body mics.

I use the Ivie analyzer to set uniform SPL in the room before I set EQ.

Over-equalization can seriously degrade the sound quality. No EQ is better than too much.

The first step is to have a correct speaker system. We've had success with Meyer Sound Labs speakers because they are consistent. Once you find a sound you like, the SIM lets us recreate that same sound from one venue to the next.

Notching out feedback frequencies is effective for a fixed installation, but not for a show on the road.

Describe The Effects Devices You Have Used And How You Use Them.

The Lexicon 480 is brilliant—fun—a lovely machine. Before that, the 224 was good, and the PCM 70.

Gating is difficult in live theater because the ambient sounds open the gates.

The Drawmer noise gate is good, as is the Brooke-Siren compressor/limiter in de-ess mode. So the dbx de-esser.

I delay the wireless mics so that the audience localizes the sounds at the actors, and I de-ess the speaker signal so that it's less distracting.

The Yamaha SPX-90 multiple-effects processor has much flexibility, and is good for lower budgets.

What Are Your Desired Features In Mixing Boards?

It's helpful to have input displays that pre-warn you of trouble before you send the sound to an audience. Polarity-reverse switches and group matrix arrays are useful.

Reliability is most important. The board also should be quiet and human engineered.

The $230,000 Cadec board is a programmable theater console, useful for long show runs. The Yamaha PM-3000 is very reliable and is easy to set up quickly. The Yamaha DMP-7 is programmable.

Besides Reliability, What Other Features Are Important In A Power Amplifier?

Resettable levels and signal status indicators.

What Are Your Criteria For Selecting Speakers?

They should be inconspicuous and have a flat frequency response. Controlled dispersion is important. Two 60-degree horns, aimed carefully, can project sound to the audience while keeping sound off the thrust stage.

The speaker should sound the same at low and high levels. Multiple horn systems should perform well at the outermost areas of their patterns.

An integrated, protected speaker system (like Meyer Sound Labs) is more reliable.

I go with JBL sometimes because they sound good at low levels.

How Do You Get The Board Operators To Do What You Tell Them?

Tell them a story to give an image of what sonic impression you want. Know your operators and work with them so that they think the same way you do.

How Do You Relate To House Personnel In A Publicly Owned Auditorium?

Listen to and respect what the house person has to say.

Seek his or her advice regarding what will or won't work.

What About Unions?

The sound designers' union 922 sets up rules that protect individuals and set salaries. Don't expect house union people to work at a breakneck pace every day—they do hard work and have to pace themselves.

In Using Delay To Move The Apparent Source Of Sound To The Actor: If You Use Too Much Delay, You Lose Intelligibility. How Many Msec Delay Do You Use?

I use 15 msec delay for an average-size proscenium arch. This localizes the sound at the actor, which helps the audience sort out what the different characters are saying, and prevents the sound system from calling attention to itself.

Does The Producer Give You A Budget, Or Do You Tell The Producer What Your Required Budget Is?

It's done both ways. If you can't do a good job with the producer's budget, and that budget is inflexible, it's better to turn down the job rather than risking your reputation by specifying an inadequate sound system. db

Jagger Down Under

It's wonderful the way fate works! There I was, lying in the sun in the backyard, wondering what the future held for a recording engineer like myself after a hectic twenty years.

Having recorded my fair share of top Aussie bands, and the fact that I had a family of seven which made a jaunt overseas next to impossible, I wondered what the future could possibly hold.

Well, the answer came with a very excited call from the Metropolis studios telling me that Mick Jagger's management was looking for an engineer to record a live television special which would include some studio work, and was I interested in submitting my work reel and bio? Was I interested!

A NEW PROJECT

Instantly the images of an early retirement vaporized and I was madly rummaging through old audio and video tapes of live recordings I had done. Should I include old stuff like Mike Nesmith live at the Palais or Sports live from Bombay Rock?

After a lot of listening, looking and discussion with the team at Metropolis Audio, it was decided to include only recent material I had done, such as Noiseworks which I recorded live at Selina's nightclub in Sydney, and the Little River Band concert which was broadcast live from the opening of World Expo 88 in Brisbane.

Well, I guess by now you've figured out that, much to my excitement, I got the gig. After the euphoria had dissipated, the team of Ern Rose, Tom Kehoe and myself set about organizing the event.

Ian "Mack" McKenzie is the Chief Engineer/Director of Metropolis Audio Pty. Ltd., South Melbourne, Australia.

The special was to have three main elements. The bulk of the special was to be videotaped at the Melbourne Tennis Centre. This venue has 360-degree seating for 15,000 and good acoustics, making it ideal for television. Then two or three songs were to be filmed at the Corner Hotel, a club in Richmond, a suburb of Melbourne. And, finally, some footage of the song "Party Doll" was to be shot at Broken Hill in outback Australia to give the special a "downunder" feeling.

After discussions with Tony Blanc, the sound man for the Mick Jagger tour, we decided to run 45 channels from the stage to the Metropolis Audio OB van. The van has a 40 channel Harrison series 4032 with two MCI 24-track recorders, so we added a Yamaha 1800 desk to handle the extra lines.

We ran the 25 drum mics, bass, and stereo keyboard feeds into the Yamaha and the vocals and guitars into the Harrison. This left enough room on the Harrison to use for monitoring only and negated the "do we use the EQ for the mic going to tape or keep it for the monitor?" problem.

We then combined these channels to 21 tracks allowing two for audience and one for time code being supplied by the video van. All inputs to the MCI recorder were split and sent to the audio booth in the video van where a rough mix could be done.

I decided to run at 30 in./sec. using Dolby A noise reduction as there was some quiet acoustic numbers in the concert. This meant I could record Jagger's vocals at quite a low level and not be concerned about tape noise or dynamics. I didn't want to use any limiting on Jagger's vocal as I was already concerned about the compression factor of the Sony radio mic he was using.

A SMALL PROBLEM

The recording of the Friday and Saturday night concerts went without any problems, except for an over-enthusiastic guitar roadie who tested a spare guitar radio transmitter while the main one was in use and they cancelled each other out, resulting in a missing acoustic guitar in the first half of "Party Doll."

We took Sunday off to listen to the rough mixes and work out the plan of attack for the following week. As it turned out, the performances on both nights were great and "War Baby" was chosen from Friday and the rest from Saturday. On Monday we arrived at Metropolis Audio studios ready to begin the post-production.

The heart of Studio One is a 6000 series SSL console. Hanging off this were a MCI 24-track machine, 2 Sony 3/4-inch U-matics for playback, a Sony 3324 digital machine, and ATR center-track timecode machine to mix down to. All these machines were connected to a sophisticated synchronizer called "Editron." The reason for two U-matics was to display the edited pictures on one monitor and the camera, which was on Jagger all night, on the other. This meant that if we had a problem with vocal quality due to the radio mic dropping out we could repair the track in the studio and keep lip synch.

The opening title's music had already been recorded in Los Angeles. The tempo was a bit slow to use for the end credits, so first up we set about recording a faster version of the same piece.

INNOVATIVE SOUNDS

While I programmed a simple drum machine pattern for the click track, Simon Phillips, the band's drummer, set about hitting everything in sight looking for that elusive rikitty tick snare sound. He found the timber floor in the third live room to be just the thing.

Meanwhile, Jimmy Rip had removed his boots and he too was hitting everything with them, looking for a kick sound. It was agreed that "cowboy

Figure 1. The legendary Mick Jagger pictured with engineer Ian "Mack" McKenzie in Metropolis Studio One during the finishing post-production touches on his television concert program for the Channel Nine Network, "Mick Jagger : Deep Down Under."

boots banging on guitar road case" made the ideal bass drum thud.

So with Simon on his knees playing floor, and Jimmy playing guitar case, the drum track was recorded. Joe Satriani and Jimmy then overdubbed some guitars and Jagger played some harmonica through a mic plugged into a Rockman on total distortion for a little rough edge.

Tuesday saw us repairing the missing guitar on "Party Doll," then starting the mix. I kept all effects in the mix to a minimum, relying more on the audience mics to provide the main ambience. Two Lexicon 480Ls, an AMS RMX16 and a couple of Korg SDE3000 delays were the only extra help needed.

To everyone's horror, when we came to mix the Broken Hill part of "Party Doll," the worst fear of any sweetening session was realized! There was no synch and the 24-track MCI kept getting further ahead of the pictures. I averted major panic by explaining that one of the features of the Editron synchronizer could save the day.

This clever piece of equipment allows you to line up the start of the picture with the start of the audio and mark it. Then you line up the end of the picture with the end of the audio, and Editron will calculate the capstan speed of the MCI to smoothly and imperceptively slew between the two points. It really is a fantastic feature and has saved many potential disasters.

MICK'S ENTHUSIASM

By Friday we were running a bit behind schedule and the band had to leave for Adelaide. Jimmy decided to stay behind and help finish the mix. Then, much to my surprise, Mick walked in and said, "Hey you guys, you can't get rid of me that easily." Mick was very much a part of the team and his tireless enthusiasm rubbed off on everybody, making the sessions a lot of fun and very rewarding.

I was also amazed at Mick and Jimmy's stamina. As 5am approached and my assistant, Doug Roberts, and I started to fade, the guys were out in the studio playing their 25th game of table tennis, and I don't mean just tapping the ball over the net. These guys were really working out!

We transferred the final mix at 7am. Jimmy and Mick flew out on a charter flight to Adelaide, and I went home to contemplate what the future held for a recording engineer like myself after a hectic twenty years and two weeks! db

Broadcast Audio

AUDIO EQUIPMENT INTERFACING: IMPEDANCE AND LEVELS

• The designer of an audio system or facility, large or small, must decide how the separate components of that facility will interface to one another. Optimal functioning of any audio system requires that when the system is constructed, careful consideration be given to such factors as electrical power distribution, grounding and shielding, and to the subject of this column, the interconnection impedances and signal levels employed.

When video equipment in interconnected, there is but one standard impedance and one standard signal level to use, but when audio equipment in interconnected, the system designer is presented with options. Professional audio interface schemes have varied through the years, depending upon the design characteristics of available equipment and the nature of the system under consideration.

TWO PHILOSOPHIES

There are two fundamental audio interconnection philosophies: power matching and voltage matching. Power matching evolved from the early days of broadcasting, and the nature of audio equipment design at that time. An example of power matching is the 600 ohm system that developed from early telephone company practice. A given piece of audio equipment has an output impedance of 600 ohms, and is connected to another piece of equipment, which has an input impedance of 600 ohms. A 600 ohm source is terminated with a 600 ohm load: a matching termination. A matching termination produces maximum power transfer, a desirable objective for equipment incorporating vacuum tube electronics and transformer coupling, but not necessarily required or desirable with today's solid state audio equipment. It also attenuates the open-circuit output voltage of the source by 6 dB, or one-half, thereby adding to the drive requirement of the output stage.

When long audio lines are used, cable capacitance becomes a significant factor, producing substantial high frequency attention in a 600 ohm system. Reducing the system impedance to 150 ohms solves the cable capacitance problem for lines of a length that might be encountered within most audio facilities, even very large systems such as those of a television network, for instance. The largest audio facility would generally not have many cables that exceed 1000 feet in length, and cables of such length present capacitive loads that may be overcome using 150 ohm power matching. 150 ohm power matching does, however, require high current drive capability from audio output stages employing it.

Standard practice for telephone company loops, which usually exceed 1000 feet, and in fact often traverse distances in excess of a mile between amplifiers, is to use 150 ohm impedance on the loop itself, translated to 600 ohms at the interface points at either end. The lengths of telephone company cables are such that cable capacitance causes high frequency roll-off even at 150 ohms. The equalizers on telco loops compensate for high frequency roll-off by attenuating the low frequencies a like amount, and the result is flat frequency response, but substantial loss in signal level, which is made up with amplification.

Telco loops are of such length that transmission line effects are exhibited at audio frequencies. Transmission line effects require impedance matching to prevent signal reflections. Signal reflections produce "ghost" images in video, while their results in audio systems are frequency response aberrations and comb filter effects. Transmission line effects may be disregarded within even a very large audio system, because transmission line characteristics do not come into play until cables reach a length of about one-tenth of a wavelength. One-tenth of a wavelength at 20 kHz. the highest audio frequency of interest, is about a mile. So while the telephone companies must be concerned about transmission line theory in audio cables, the designer of any conceivable audio system within a single location may disregard such effects.

The second fundamental interconnection philosophy is voltage matching. Voltage matching employs a low output source impedance, typically 50 to 100 ohms, and terminating impedances that are sufficiently high to be considered "bridging" with respect to the source. Input impedances for this type of equipment commonly range from 10 to 50 kilohms. In a voltage matched system the open-circuit voltage of an output stage is attenuated only very slightly, if at all, when the load is connected. Contemporary solid state electronic design has produced virtually universal compatibility with the voltage source concepts of low-impedance outputs and high-impedance inputs, and this is true of much transformer-output equipment as well as transformerless.

Along with the two impedance philosophies, standard audio signal levels vary as well. The audio level traditionally used in a 600 ohm power matched system is +8 dBm. The term "dBm" specifies a ratio relative to a reference level of one milliwatt, in any impedance. One milliwatt is a *power* level, and a given power level produces different voltages across different load impedances. Thus 0 dBm, one milliwatt, produces 0.775 V_{rms} across 600 ohms, or 0.387 V_{rms} across 150 ohms. Translating some common reference levels to voltages, +8 dBm/600 ohms = 1.95 V_{rms} + 10 dBm/150 ohms - 1.224 V_{rms} and +4 dBm/600 ohms = 1.228 V_{rms}. Voltage matching introduces another unit of audio level measurement, the "dBu" specifies a ratio with respect to a reference *voltage* of 0.775 V_{rms} without regard to impedance. The most common reference level used in voltage matched systems is +4 dBu, or 1.228 V_{rms}.

SYSTEM PREFERENCE

The power matched systems still in use are relics of the vacuum tube era, and are anachronisms in the 1980s. It is becoming standard practice all over the world to use voltage matched audio interfacing. Here are some specific reasons for preference of voltage matching to power matching:

1) A substantial increase in bandwidth is realized on a system level. The bandwidth limiting factor in an audio system employing modern solid state equipment is not the electronics, but the cable.

2) Interconnection and termination is faster and easier because there are no termination resistors to deal with. Level problems caused by missing or double terminations are eliminated.

3) Smaller signal current flow in cables, reducing the potential for crosstalk.

4) Capacitive coupling between cables is reduced because of the lower impedance held on lines.

5) Multiple loads may be fed from the same source if necessary. Ten, 10 kilohm loads still present a bridging impedance to a 50 ohm source. Such multing is not generally recommended as a system wiring practice because of reliability considerations, but may be used when patching.

6) Lower system distortion results from the reduced loading factor. The greater the load on an output stage, the more distortion it generates. In particular, driving a 150 ohm load taxes output stages to their utmost, and requires far more drive current than driving a high-impedance load.

7) Greater headroom and thus wider dynamic range results from driving high impedance loads.

8) Voltage matched systems consume less power than power matched systems, and thereby reduce power supply requirements.

Frequency response sweeps of circuits representing the common interconnection schemes illustrate the increased system bandwidth available when voltage matching is used. First, consider a system consisting of a 600 ohm source feeding a 600 ohm load via 1000 feet of twisted-pair cable with shield, representative of audio cable in common use today. The capacitance between signal-carrying conductors of such cable is around 30 picofarads per foot. This capacitance creates a low pass filter, and for the 600 ohm system this filter's –3 dB point falls at about 14 kHz, with the response at 20 kHz being rolled off about 5 dB. Perceptible roll-off begins at frequencies as low as 4 kHz.

A sweep of a similar system with source and load impedance reduced to 150 ohms reveals frequency response flat to 20 kHz with the –3dB point falling above 30 kHz. The reason for choosing 150 ohm systems over 600 ohm systems for long cable runs is obvious. The 150 ohm power matched system provides sufficient system bandwidth, even with 1000 foot cables. However, at 30 pf per foot, 1000 feet of cable presents a capacitive load of 0.03 pf, which appears in parallel with the load resistance, further taxing the output stage and generating more distortion. When the source impedance is reduced to 50 ohms and the load impedance is raised to 10 kilohms, the –3 dB point is well above 100 kHz. The cable capacitance appears in parallel with the much higher load resistance, and is driven by the low impedance driver, greatly reducing the demands on the output stage.

It may appear that the lower the source impedance the better, but in fact if the driving impedance is lowered sufficiently, high-frequency peaking will be seen. The optimum source impedance for voltage matched systems is about 60 ohms, and the 50 to 100 ohms found in typical audio equipment is perfectly satisfactory.

When all the above is considered, it is not difficult to reach the conclusion that voltage matching is preferable to power matching for interconnection of today's audio systems.

STANDARD LEVEL

The common reference level for voltage matched systems is +4 dBu, or 1228 V_{rms}. Virtually all modern professional audio equipment is compatible with this level, which is in fact the standard level used in recording studios, and increasingly, in broadcast audio systems. It is 4 decibels lower than the standard telephone company level of +8dBm, which is the reference level traditionally used in 600 ohm broadcast systems. The noise floor of all modern professional audio equipment is sufficiently low that excellent dynamic range may be achieved using a reference level of +4 dBu. Most audio electronics of recent vintage is capable of at least +24 dBu undistorted output, providing 20 dB of headroom above reference level.

Some audio facilities require subsystems designed for power-matched operation. For example, telephone companies often use passive equalizers which require a true 600 ohm source (not a low impedance source capable of driving a 600 ohm load) to produce proper frequency response.

With the above exceptions noted, voltage matched audio interfacing is the interconnection philosophy of choice, and +4 dBu = 1.228 V_{rms} has become reference level of choice for professional analog audio interfacing. Implementing such a philosophy is relatively straightforward for equipment with balanced transformerless input and output stages, which are rapidly becoming the rule rather than the exception. Equipment with transformer inputs or outputs may fall into several categories. True 600 or 150 ohm outputs may be source-terminated in their characteristic resistance, making them appear as voltage sources to the outside world. Transformer outputs which are actually low-impedance sources, for instance, transformers designed to drive 600 ohm loads, but having an actual output impedance around 100 ohms, often may be used without terminations. Such equipment should be checked for frequency response in the unterminated mode before being used in this way, However, because termination may be required for proper frequency response.

Direct connection of multiple loads to a single source is permissible when voltage matching is used. This is not recommended as a distribution practice, although it may be used when patching. Connecting multiple loads to a single output can compromise reliability, because if one output fails or becomes shorted, all loads are affected. It is prudent therefore, to isolate loads with multiple distribution amplifier outputs. If audio distribution amplifiers use 60 ohm balanced output splits, each output is protected with about 60 dB of isolation against a backfeed from any other output, and shorting any output results in a negligible voltage drop on other outputs.

It is apparent that there are many advantages to using voltage matching interface techniques in audio system design. Such design will result in improved technical operation and reduced power consumption. dB

1989 Editorial Calendar

JAN/FEB .... db Looks at the Electronic Cottage going upscale!
A broadcast report on the Seoul Olympics
• GUIDE: Speakers: performance & monitor

MAR/APR .... db Looks at the Sound Reinforcement Scene: theory, layout, and construction
• GUIDE: Power Amplifiers

MAY/JUNE .... db Looks at The Windy City
• GUIDE: Consoles & Mixers

JULY/AUG .... db goes on tour with the Major Touring companies
• GUIDE: Tape, tape recorders and accessories
Microphones

SEPT/OCT .... db looks at the Boston Recording Scene
• GUIDE: Signal Processing Equipment, Part I

NOV/DEC .... db Looks at The West Coast & Hawaii
• GUIDE: Signal Processing Equipment, Part II
Studio Accessories

BUYER'S GUIDE:

Power Amplifiers

Introduction to the Charts

We've tried to make the charts of amplifiers as self-explanatory as possible, with slanting headlines on each column that explain what we wanted to show you.

These charts represent entirely what each of the respective manufacturers have sent us in response to our (sometime repeated) requests. You will also see that there are numbers of blank sections within the charts. If they don't have a specification available, we can't list it. But note that many do not have anything under the Features column. This column is where we have invited each manufacturer to state, in as few words as possible, what is special about the product. You can safely assume, then, that when this column is blank, it is because the manufacturers told us nothing.

Note also that we ask for amplifier continuous power not only at the traditional 8 and 4 ohm resistive loads, but also at 2 ohms. As you know, when you parallel speakers, the load is halved. Accordingly, in the real worlds of studio monitors and headphone lines, and the even more real world of performance and stadium systems, effective loads back to an amplifier can well be 2 or 3 ohms. Since modern solid-state amplifiers can handle such loads successfully, we ask each manufacturer for this specification. Note that not all give it. It's therefore safe to assume that if it is missing, the amplifier may not be reliable at low loads.

Distortion at normal and full power ratings is also specified. While many amplifiers today can boast of almost vanishing distortion, remember that if you will be pushing an amplifier hard up against its rated power and beyond, distortion will then be rising rapidly. No audio product is really made to be abused, and amplifiers are no exception.

One group of important specifications deals with dimensions and weights. Amplifiers, particularly high-power ones, are not lightweights. A few racks can have weights adding up rapidly.

Finally, the price. What we have asked each manufacturer for is the suggested retail price. Different retail dealers establish their own.

On to the charts...

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
ALTEC LANSING CORPORATION															
1268	2	60	100		20-20k	.03	.03	.3	.1	5-100k	.78	3.5 19 10	33	$1330.00	Includes choice of XLR, ¼-inch, and terminal inputs, 5-way binding posts, LEDs, protection.
1269	2	120	200		20-20k	.03	.03	.3	.1	5-100k	.78	3.5 19 14.75	47	$1430.00	Choice of XLR, ¼-inch and terminal inputs, 5-way binding posts, peak and protection LEDs.
1270C	2	220	400		20-20k	.05	.05	.3	.1	7-33k	.78	5.25 19 15	51.5	$1850.00	Choice of XLR, ¼-inch and terminal inputs, peak and protection LEDs, two-speed fan cooling.
1407A	1	75	75		20-20k	.01	.01	.2	.1	10-30k	.78	5.25 19 12.5	24.2	$556.00	Includes XLR (male and female), phono and terminal inputs, direct and X-former output, protection.
1415A	1	150	150		20-20k	.01	.01	.2	.1	10-30k	.78	5.25 19 12.5	30.8	$680.00	XLR (male and female), phono and terminal inputs, driver protection relay, convection cooling.
2200A	var				20-15k			.5	.25	10-30k	.81	7 19 17.4	70	$5100.00	Eight 75W power amp modules that can be configured for numerous output combinations up to 600W.
2204A	4	75 (4ch)	150 (2ch)	300 (1ch)	20-15k			.5	.25	10-30k	.78	5.25 19 16	31.5	$1750.00	Four 8 ohm/75W power outputs that can be paralleled and/or bridged, X-former isolated inputs.
9444A	2	200	300		10-70k	.05	.05	.2	.1	10-85k	.81	5.25 19 12.75	39	$958.00	Electronically balanced XLR or terminal inputs, octal socket, peak and protection LEDs.
ASHLY AUDIO INC.															
FET-200	2	120	190	225	20-20k	.004	.01	.004	.05	20-20k	1.4	3.5 19 16	37		Modular construction forced air cooling, LED meters.
FET-500	2	300	500	675	20-20k	.004	.01	.004	.05	20-20k	1.7	5.25 19 16	60		Stable into any load mono/bridge modes, brown-out proof.
BGW SYSTEMS INC.															
7500T	2	200	250		20-20k		.05			3-100k	1.22	5.25 17.5 10.5	36		Modular construction barrier-strip terminal, plug-in crossover.
SPA-1	2	250	400	600	20-20k				7.05			5.25 19 13.7	41		Signal processing subwoofer amp parametric EQ.
SPA-3	2	250	400		20-20k				7.05			5.25 19 13.7	43		Signal processing amplifier, active, balanced inputs.
750FG	2	280	450		20-20k		.01		.06	20-20k	1.5	7 17.3 12.9	55		Low feedback design, available in a studio version.
GTA	2	350	600	900	20-20k		.03			20-20k	1.48	7 17.5 14.5	72		Grand touring amp, solid state DC speaker protection.
GTB	2	275	400	800	20-20k		.03		.1	20-20k	1.48	5.25 19 13	50		Accepts 2 BGW crossover cards for bi or triamping.
8500T	2	300	450	850	20-20k		.05			20-20k	1.6	5.25 19 13.8	50		Cost-effective version of GTB designed for fixed installation.
CARVIN CORPORATION —See our ad on page 15															
FET 400	2	100	200		20-20k			.006	.1	20-20k	1	5.25 19 10	31	$499.00	MOSFET technology, speaker protion, road-worthy construction.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
FET 900	2	200	300	450	20- 20k			.006	.1	20- 20k	1	5.25 19 10	35	$669.00	Same features as the FET 400, yet has 900 watts total output.
CREST AUDIO															
1001A	2	55	85		20- 20k	.04	.04	.06	.06	20- 20k	.42	10.5 19 1.75	17	$679.00	
1501A	2	90	150		20- 20k	.04	.04	.06	.06	20- 20k	.61	10.5 19 1.75	17	$839.00	
3000-1	2	260	475	720	20- 20k	.01	.01	.03	.03	1- 50k	1.1	5.25 19 11.5	46	$1589.00	Includes signal present LEDs.
4000-1	2	340	600	900	20- 20k	.01	.01	.03	.03	1- 50k	1.27	5.25 19 13	58	$2189.00	Includes signal present LEDs.
7001	2	560	810	850	20- 20k	.01	.01	.05	.05	20- 20k	1.4	3.5 19 15	49.5	$2589.00	
8001	2	750	1225	1400	20- 20k	.015	.015	.05	.05	20- 20k	1.75	5.5 19 15	80	$3189.00	
PL-400	2	320	500		20- 20k	.01	.01	.01	.1	20- 20k	1.2	3.5 19 13	38	$1239.00	
FA-800	2	240	400		20- 20k	.019	.019	.01	.1	20- 20k	1.0	3.5 19 13	32	$879.00	
CROWN INTERNATIONAL															
Macro Tech 600	2	235	325	340	35k	.05	.05	.001	.05	20- 20k	.77	3.5 19 16	43	$1295.00	
Macro Tech 2400	2	525	800	1200	35k	.05	.05	.001	.05	20- 20k	.77	3.5 19 16	60	$1995.00	
Micro Tech 1200	2	320	465	600	35k	.05	.05	.001	.05	20- 20k	.77	3.5 19 16	45	$1295.00	
D-75	2	35	45	60	35k	.01	.01	.001	.05	20- 20k	.81	1.7 19 9	13	$524.00	
D-150A	2	95	150	108	35k	.01	.01	.001	.05	1- 20k	1.19	5.2 19 8.7	27	$795.00	
DC-300A	2	175	295	320	35k	.01	.01	.001	.05	0- 20k	1.75	7 19 9.7	45	$1095.00	
PSA-2X	2	265	380	685	50k	.01	.01	.001	.05	20- 20k	2.10	7 19 14.7	61	$1995.00	
PS-400	2	165	265	225	35k	.01	.01	.001	.05	0- 20k	1.76	7 19 10	54	$1019.00	
dbx, INC.															
BX1	2,3 or 4	100 (4-ch.)	200	325 20k	20-	.01		.01	20k	20-	1 19	7 24.5	84	$2800.00	Reference quality amp, configurable for two, three, or four channel operation.
ELECTRO-VOICE, INC. —See our ad on page 7															
AP2600	[illegible]	200	300				.03		.05		.775	5.25	39	$999.00	Available with precision stepped

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
					85k							19 12.75			attenuators as model AP2600SA.
HAFLER															
P500	2	225	400		10- 60k	.007	.007	.007	.002	20- 24k	1.55	7 19 13	53		Balanced line, mono bridgeable, front level controls.
P505	2	225	400		10- 60k	.007	.007	.007	.002	20- 24k	2.55	7 19 13	48		Mono bridgeable, level controls.
P230	2	115	175		10- 60k	.005	.005	.007	.002	20- 24k	2.55	5.25 19 10.5	28		Balanced line accessory, mono bridgeable, level controls.
P125	2	62			4- 200k	.005	.005	.007	.002	10- 40k	1.1	3.5 19 9	22		Balanced line accessory, mono bridgeable, level controls
INDUSTRIAL RESEARCH PRODUCTS, INC.															
DH4020	2	100	140		20- 20k			.02	.1		1	1.75 19 14	13.5	$1042.00	100 kHz switching power supply, MOSFET high freq. output.
INDUSTRIAL STRENGTH INDUSTRIES —See our ad on page 23															
PA700	2	250	400		10- 50k							5.2 19 13.2	40	$699.00	Built-in electronic X-over, balanced XLR inputs, mono bridging, protection, automatic fan cooling
INNOVATIVE ELECTRONIC DESIGNS, INC.															
6208	1	200			20- 20k			.5	.5	20- 20k	.9	6.1 2.1 12.7	2.4	$783.00	Class D (switching mode) design card type, greater than 80% efficiency at full output.
6270	1				20- 20k			.5	.5	20- 20k	.9	6.1 2.1 12.7	2.38	$783.00	200 watts into 25 ohms or 70.7v, all specs at 25 ohm load. Both amps have balanced outputs.
JBL PROFESSIONAL															
6210	1	40			20- 20k	.1	.1	.1	.1	20- 20k		8 8.5 2.75	6.5	$295.00	Converts 4400 series or any other 8-ohm monitor into self-contained power system.
6211	1	40			20- 20k	.1	.1	.1	.1	20- 20k		8 8.5 2.75	6.5	$330.00	Same as above and also includes active balanced inputs, but with mic/line selector switch.
6215	2	35	45		20- 20k	.1	.1	1	1	20- 20k	1.1	1.75 19 9	10.5	$650.00	Active balanced bridging input circuitry, rear-panel switch for bridge/dual mono or stereo.
6230	2	75	150		20- 20k	.1	.1	1	1	20- 20k	1.1	5.25 19 11	26.3	$695.00	Same features as above.
6260	2	150	300		20- 20k	.1	.1	1	1	20- 20k	1.1	7 19 11	44.5	$995.00	Same features as above.
6290	2	300	600		20- 20k	.1	.1	1	1	20- 20k	1.1	7 19 14	63	$1495.00	Same as above yet also includes fan for cooling.
PASO SOUND PRODUCTS															
Alpha 130	2			200	10- 40k			.003		5- 100k	1	5.33 19 13.5	28.5	$376.00	
Alpha 230	2			245	5- 100k			.008		5- 100k	1	5.33 19 13.5	29.3	$485.00	
Alpha 450	2			440	5- 100k			.008		5- 100k	1	7.25 19 18.13		$899.00	
Alpha 650	2			680	10-			.008		0-	1	7.25	60.5	$1485.00	

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
					40k					100k		19 18.5			
PEAVEY ELECTRONICS															
CS-1200	2	350	600		20-40k		.01		.03	5-60k	1.4	7 19 17.75	70	$1299.99	Plug-in electronic X-over capability, transient-free turn on/off, thermal protection.
CS-1000	2	300	500		20-40k		.01		.03	5-60k	1.4	5.25 19 14.13	50	$999.00	Plug-in electronic X-over capability, transient-free turn on/off, thermal protection.
CS-800	2	240	400		20-20k		.04		.03	5-40k	1.4	7 19 13.5	54	$799.99	Plug-in electronic X-over capability, DDT compression, fan-cooled, thermal protection.
DECA 528	2	210	250		20-20k	0.0	0.0		.15	10-20k	1.0	1.75 19 14	12	$749.99	Digital energy comversion amp, DDT compression, fan-cooled, MOSFET design.
DECA 1200	2	325	600		20-20k	0.0	0.0		.06	10-20k	1.3	3.88 19 18	37	$1399.99	Digital energy comversion amp, DDT compression, fan-cooled, MOSFET design.
DECA 724	2	225	350		20-20k	0.0	0.0		.1	10-20k	1.0	3.88 19 18	37	$999.99	Digital energy comversion amp, DDT compression, fan-cooled, MOSFET design.
M-7000	2	200	350		20-20k		.01		.03	10-30k	1.4	5.25 19 16.5	47	$749.99	Fan-cooled, DDT compression, electronic X-over capability, transient-free turn on/off.
QSC AUDIO PRODUCTS, INC.															
1100	2	50	70	90	40k		.01	.01	.01	20-20k	.83	1.75 19 7	12	$568.00	
1400	2	200	300		40k		.01	.01	.01	20-20k	1	5.25 19 9.5	34	$798.00	
1700	2	325	500		40k		.01	.01	.01	20-20k	1	7 19 10.8	57	$1098.00	
MX1500	2	330	500		40k		.01	.01	.01	20-20k	1	3.5 19 17.9	47	$1098.00	
MX2000	2	375	625		40k		.01	.01	.01	20-20k	1	5.25 19 15.9	75	$1498.00	
3200	2	110	140		40k		.01	.01	.01	20-20k	1	1.75 19 14.6	26	$958.00	
3500	2	300	450		40k		.01	.01	.01	20-20k	1	3.5 19 15.9	50	$1488.00	
3800	2	375	600		40k		.01	.01	.01	20-20k	1	5.25 19 15.9	75	$1938.00	
RAMSA (PANASONIC)															
WP9440	2	350			10-60k	.06	.06	.06	.06	20-20k	1.23	5.25 19 18.88	75	$2090.00	Intelligent VI limiting, soft overload characteristics.
WP9220	2	200	300		10-85k	.06	.06	.06	.06	20-20k	1.23	5.25 18.88 15.06	38.6	$1090.00	Has ability to drive high phase-angle, loads with ease.
WP9110	2	100	150		10-85k	.06	.06	.06	.06	20-20k	1.23	3.5 18.88 15.06	28.6	$840.00	Detented input attenuators with removable knobs.
WP9055	2	50			10-85k	.05	.05	.05	.05	20-20k	1.23	1.75 18.88 13.13	19	$590.00	Signal, peak and protect LEDs, XLR and phone inputs.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
RANE CORPORATION															
MA6	6	100	150		20-20k	.2			.2	5-80k	.775	5.25 19 11	44	$1349.00	Built-in limiters, auto bridging, 2-speed fan.
RENKUS-HEINZ, INC.															
P-1500	2	300	500	750			.1		.1	20-20k	1.55	3.5 19 13.75	40	$1750.00	Compact, rugged amp, features Power-Guard circuitry.
SHURE BROTHERS INC. —See our ad on Cover IV															
210	1	6	10		100-15k			1	3	100-15k	40mV	2.75 9.5 5.63	2.13	$125.00	Balanced mic input, unbalanced line in-put, ext. 12V power.
SOUNDCRAFTSMEN —See our ad on page 37															
300X4	2,3 or 4	600 205	900 300	450	20-20k	.05	.05	.008	.05	20-20k	1.0	5.25 19 14	60	$1299.00	Multi-channel MOSFET, 2, 3 or 4 channel mode indicators, front panel-mounted circuit breakers.
PM860	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k	1.2	5 8.5 14	20	$599.00	Has high current design to allow stability with 2 ohm loads.
450X2	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k	1.2	5.25 19 11.75	28	$849.00	High current MOSFET amp with balanced or unbalanced inputs.
900X2	2	375	675	900	20-20k	.05	.05	.008	.05	20-20k	1.22	5.25 19 16.5	59	$1599.00	Same as above.
RA7501	2	275	420	320	20-20k	.05	.05	.05	.05	20-20k	1.21	7 19 15	47	$949.00	Class H signal tracking design for maximum efficiency.
SPECTRA SONICS															
701	1-Inf	33	58	86	20-20k	.05	.075	.025	.025	0-20k	+5 dBv	2.5 10 1.88	.88	$108.00	A modular amp suited for bi, tri, multi-way, used in noise masking, broadcast, recording.
701BP	1-Inf	122	172	200	20-20k	.05	.075	.025	.025	0-20k	+5 dBv	5 10 1.88	1.76	$216.00	Is two model 701s bridged together with the same qualifications.
712B	2	30	50	80	20-20k	.05	.075	.025	.025	0-20k	0 dBv	5.5 19 14.5	22	$595.00	A stereo rack-mount, self-contained power amplifier.
712	2	100	100	100	20-20k	.05	.075	.025	.025	0-20k	0 dBv	5.5 19 14.5	24	$760.00	A stereo rack-mount, self-contained power amplifier.
SUNN															
SPL7000	2	200	350		10-50k	.005	.005	.1	.1	20-20k	1.23	5.25 19 15	40	$799.00	Quiet, 2-speed fan, delay on/off, protection, compressor.
SPL7250	2	160	250		10-50k	.005	.005	.1	.1	20-20k	1.23	3.5 19 14.5	30	$599.00	Balanced ¼-inch/XLR, 5-way binding posts.
STUDER REVOX AMERICA, INC.															
A68	2	100	175		30-20k				.1		17dB	5.25 19 9.75	48.4	$2100.00	Low closed loop feedback, mono bridgeable for 400W.
B242	2	200	230	400	20-20k		.01		.01	20-20k	1.55	6 14.2 7.7	37.5	$3000.00	MOSFET drive transistors, 2 separate power transformers.
SYMETRIX															
A-220	2	20	20		20-40k				.01	20-20k	.5	1.75 19 8	9	$315.00	Phone and XLR input, front panel headphone jack.
YAMAHA CORPORATION															
PD2500	2	250	360	500	20-		.007		.007	10-	1.23	3.88	26.5	$1545.00	

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
					50k					50k		18.88 18.88			
P2250	2	170	250		10- 50k		.007		.007	10- 50k	1.23	5.25 18.88 18.88	41.8	$895.00	
P2150	2	100	150		10- 50k		.01		.007	10- 50k	1.23	5.25 18.88 16.63	37.4	$695.00	XLR and ¼-inch input jacks, binding post and ¼-inch output jacks, forced air cooling.
P1250	1	170	250		10- 100k		.01		.004	10- 50k	1.23	5.25 18.88 16.63	33	$595.00	XLR and ¼-inch input jacks, binding post and ¼-inch output jacks, forced air cooling.
P2075	2	50	75		10- 50k		.05		.003	10- 50k	1.23	3.88 18.88 14.38	19.8	$495.00	XLR and ¼-inch jacks, binding post and ¼-inch output jacks, compact and light-weight.
P1150	1	100	150		10- 100k		.005		.05	10- 50k	1.23	5.25 18.88 16.63	28.6	$495.00	XLR and .25-inch Input jacks, binding post and .25-inch output jacks, forced air cooling.
PC1602	2	160	240				.01		.015	10- 50k	1.23	5.5 18.88 17	47.8	$995.00	Comprehensive protection circuitry XLR in and through connectors, 480 watts in mono.
PC2602	2	260	400				.007		.015	10- 50k	1.23	7.25 18.88 17	57	$1295.00	Protection circuitry, XLR in and through connectors, model PC2602M has 26-segment backlit LCD meters

ADDRESSES

Altec Lansing Corporation
PO Box 26105
Oklahoma City, OK 73126

Ashly Audio
100 Fernwood Ave
Rochester, NY 14621

BGW Systems
13130 S. Yukon Ave
Hawthorne, CA 90250

Carvin Corporation
1155 Industrial Ave
Escondido, CA 92025

Crest Audio
150 Florence Ave
Hawthorne, NJ 07506

Crown International
1718 Mishawaka Rd
Elkhart, IN 46517

dbx, Inc.
71 Chapel St
Newton, MA 02195

Electro-Voice, Inc.
600 Cecil St
Buchanan, MI 49107

Hafler
613 S. Rockford Dr
Tempe, AZ 85281

Industrial Research
Products, Inc.
321 Bond St
Elk Grove Village, IL 60007

Industrial Strength Industries
13042 Moore St
Cerritos, CA 90701

Innovative Electronic Designs
9701 Taylorsville Rd
Louisville, KY 40224

JBL Professional
8500 Balboa Blvd
Northridge, CA 91329

Paso Sound Products
14 First St
Pelham, NY 10803-1401

Peavey Electronics
711 A St
Meridian, MS 39301

QSC Audio Products
1926 Placentia Ave
Costa Mesa, CA 92627

Ramsa (Panasonic)
550 Katella Ave
Cypress, CA 90630

Rane Corporation
10802 47th Ave W
Everett, WA 98204-3400

Renkus-Heinz
17191 Armstrong Ave
Irvine, CA 92714

Shure Brothers
222 Hartrey
Evanston, IL 60204

Soundcraftsmen
2200 So. Ritchey
Santa Ana, CA 92705

Spectra Sonics
3750 Airport Rd
Ogden, UT 84405

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

Studer Revox America
1425 Elm Hill Pike
Nashville, TN 37210

Symetrix
4211 24th Ave W
Seattle, WA 98199

Yamaha Corporation
PO Box 6600
Buena Park, CA 90622

New Products

QUAD BOX

• The Mini Quad Box is a compact quad box for wireless microphone systems. It is designed to house and centrally power up to four Vega R33 miniature receivers. It also provides RF feeds via a single antenna by means of an internal multi-coupling system. It operates from 12 internal "C" batteries or from external 12-24 volts d.c. Dimensions are 5x7x8, and weight is 7 pounds, 4 ounces.

Mfr.- Professional Sound Corporation

Price- $1275.00

Circle 60 on Reader Service Card

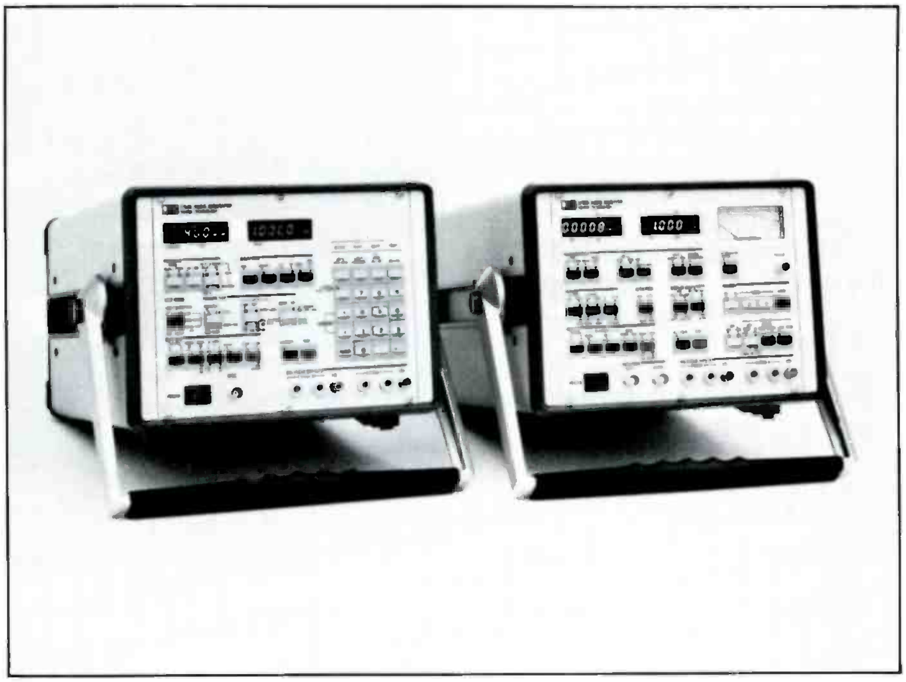
PROGRAMMABLE TEST GEAR

• The 3100B and the 3200B are programmable audio signal generator/analyzers. Designed to exceed test equipment requirements in the 16-bit digital era, these units offer three distinct modes of operation: manual with front-panel control for troubleshooting or design, built-in automation for quick equipment proofs, and complete computerized test and measurement capability using external, menu-driven software. They are designed to be portable. This two-channel unified system will manually or automatically measure operating level, frequency response, total harmonic distortion (THD) vs. level or frequency, SMPTE intermodulation distortion (IMD), signal-to-noise ratio (SNR), quantizing noise, channel separation, phase error between channels, and wow and flutter measurements. Front-panel switches, LEDs, and digital readouts on each allow usage as stand-alone units. When separated at remote sites, the generator sends FSK signals over the audio channel being tested which configures the analyzer for each test. Either or both units can be completely controlled and interrogated by a compatible terminal or personal computer (PC). The 3100B generator uses digitally-controlled analog oscillators to produce sine and square wave signals in addition to SMPTE IMD, tone-burst, and sine/step waveforms. Sine waves range from 1 Hz to 102.39 kHz, with low 0.0008 percent distortion, and square waves from 1 Hz to 50 kHz, with a rise time of less than 1 μs. Internal crystals ensure that frequency accuracy is within 0.03 percent for fixed frequency and selected frequency sweeps. The generator will store 91 different front-panel test set-ups. Whether testing in a manual or automatic mode, recalling a panel set-up speeds the procedure and ensures repeatable test conditions.

Mfr.- Sound Technology

Price- $4250.00 (3100B)
$5495.00 (3200B)

Circle 61 on Reader Service Card



MATRIX MIXDOWN

• The Model 1010MM is a matrix mix down and monitor audio panel. The specifications include 10 inputs with electronic balanced transformer, a 20 K impedance, and XL type connectors. The unit has a three-stage system which includes the input stage with audio line amplifier used to drive all ten 10 K audio taper pots. The summing stage contains an audio line amplifier of the same type used to sum all ten signals in Channel A. The output stage has ten gain controlled transformer balanced audio line amplifiers with XL type connectors. The unit's output is +24 dbm, with 0.25 percent THD, and –74 dbm noise below +4 dbm. The power supply is dual 523R (regulated ±24V at +11 amp). One supply for operation and the second for back-up in case first supply should fail (switchable). The size is 15-3/4x19x9, and the weight is 29 lbs.

Mfr. - Opamp Labs, Inc.
Price - $4500.00

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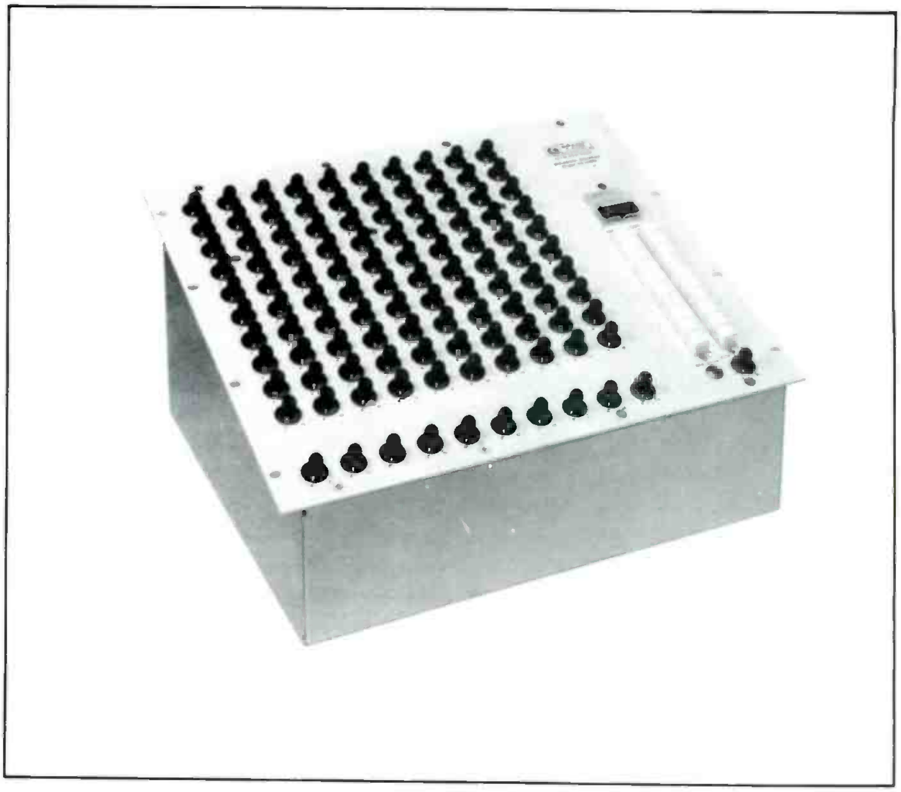
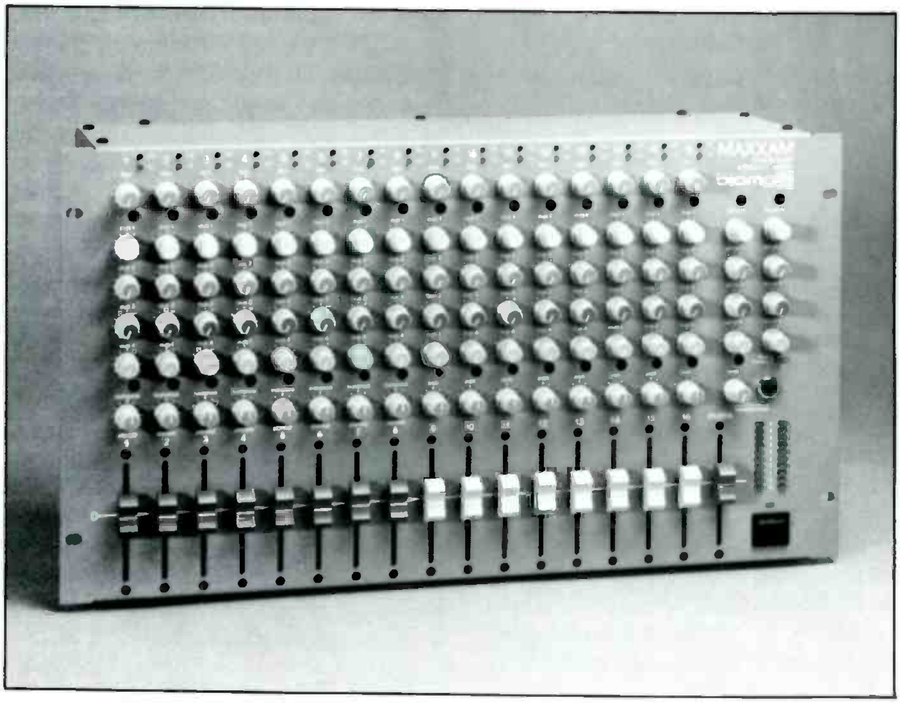


RACK-MOUNT MIXER

• The MAXXAM 8 + 8 is a rack-mount mixer with 16 input channels in a compact rack-mount configuration 6 rack spaces high. Eight of the unit's channels are designed to handle stereo input signals, giving it the input capability of a conventional 24-channel mixer. The gain structure for all channels is designed for line level inputs. The mixer includes two unassigned low noise microphone preamps with XLR input connectors and 1/4-inch phone assign jacks, thereby allowing the mic preamp output signals to be assigned to any of the input channels with a short patch cord. All input channels have patch insert jacks. The unit has four sends, one of which is selectable pre/post fader on each channel. There are also four stereo returns. Channel faders and the stereo mains fader are 60 mm throw professional level. The absence of channel EQ and use of 5532 and TL072 op amps throughout result in a very low noise unit.



Mfr. - Biamp Systems, Inc.
Price - $1499.00

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COAX MOUNTING SYSTEM

• Built to address the needs of the professional sound contractor and sound installation market, the M4 CoAx allows any of the company's PC 400 Series high-frequency horns to be coaxially mounted directly in the center of a matching M4 PC 1500 Series horn with a PCMX mouth extension. Cluster design can be simplified and reduced in size because the high-frequency horn can be located in the middle of the mid-range horn, and improved directivity performance can be obtained at the mid-to-high crossover point. If the crossover and delays are designed properly, it's almost as if there is one speaker that is capable of going from 200 to 20,000 Hz. It is available as a complete system with horns included, or as a package including all mounting hardware and a special bracket.

Mfr. - Community Light & Sound
Price - starts from $2215.00 (with horn and driver)

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REFERENCE CHART

• The Milli-Chart is a wall or table-top plastic-coated reference chart showing the relationship between digital-delay time settings and musical tempos. It was designed to alleviate the tedious process of computing delay times with a stop watch and calculator. Since echos, delays and other effects should be timed, in milli-seconds, to the beat, the chart is valuable to anyone who owns a delay, digital reverb, or multi-effects unit. It can also be used to compute tempos and the number of beats within a fixed time frame, such as a 30 second commercial. The chart gives delay times, in milliseconds, for tempos from 60 to 250 beats per minute for 1/4-, 1/8-, and 1/16-notes, 1/4- and 1/8-note triplets and beats per second. The chart is plastic-coated and measures 10x17.

Mfr. - J.S. LoBianco Enterprises
Price - $15.95

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SUBWOOFER SYSTEM

• The Model SUB-152 bandpass subwoofer system is designed to be used as sub-bass support for full-range speaker systems. By restricting bandwidth, the bandpass design increases efficiency to 106 dB (1w, 1M). Distortion components are attenuated, resulting in cleaner, tighter bass. Two high-efficiency 15-inch woofers in a specially tuned 6 cu. ft. enclosure give solid low-frequency response down to 45 Hz. The unit is for use where high-level, low-frequency energy is required. Finish is heavy-duty black carpet; painted or unfinished versions are optional. Standard connectors are dual banana and Neutrik NL4MP, and a terminal strip option can be ordered.

Mfr. - Renkus-Heinz, Inc.
Price - $1700.00

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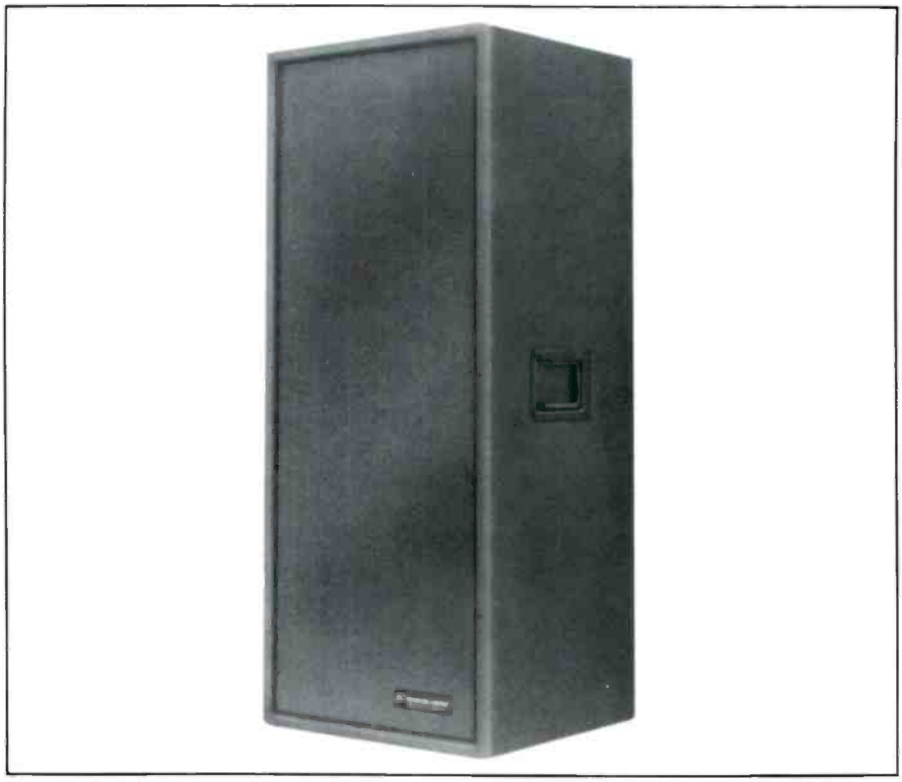
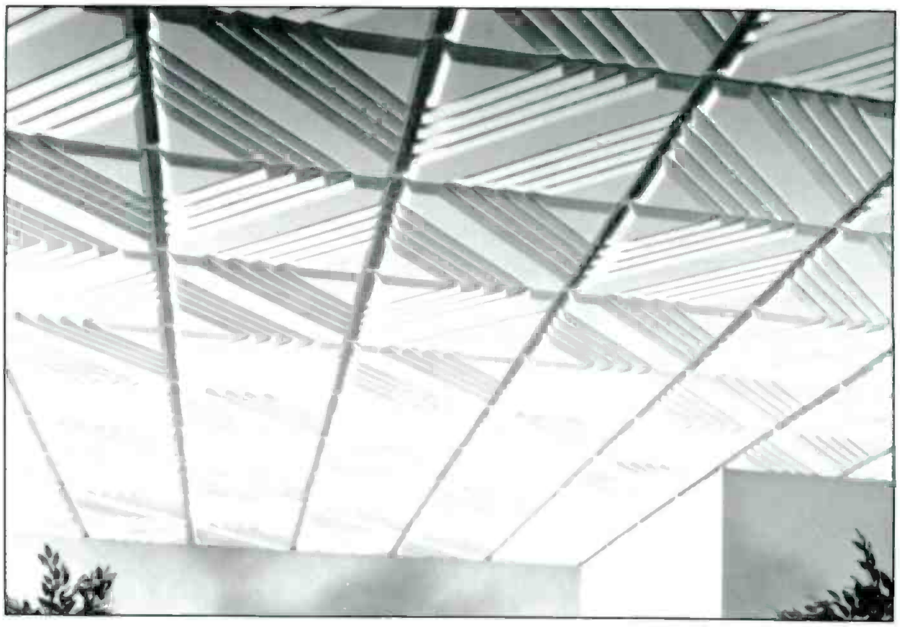


ACOUSTIC TILES

• Panorama is one of a series of five ceiling tile patterns. These SONEX ceilings are made of a lightweight, open-celled material. The deep wedges of the pattern help to deflect and dissipate sound energy while directing it into the open cells of the tile. Then the offending noise is literally trapped in the tiles' microscopic pockets. According to testing in strict accordance with ASTM C 423-84a procedures, the weighted noise reduction coefficient (NRC), which measures acoustic performance, ranged from 0.75 to 0.85. By comparison, most standard acoustical ceilings have a NRC range of 0.45 to 0.55. Made of an exclusive melamine resin, these ceilings meet all public and commercial building codes for fire safety. Each lightweight 24-inch by 24-inch by 2-inch tile fits all standard and fine line suspension grids. The soft, flexible design makes it easy to trim or cut.

Mfr.- Illbruck



Price- $5.00 or $6.00 per square foot, depending on quantity

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MONITORS

• PBM 6.5 and the PBM 8 are two playback monitors. At 12 lbs. each (11-1/2x8x8-1/2), the 6.5 is a small, cost-effective, full bandwidth mini monitor. Its rear-firing tuned port produces a low-frequency response down to 57 Hz, and allows for an extremely compact size. Its 3/4-inch ferro-fluid-cooled polymide dome tweeter extends top end high-frequency response to beyond 20 kHz. The driver accompaniment is knitted together by means of a precision hardwired crossover unit, utilizing robust low loss components, and heavy-duty input terminals which will accept standard ¾-inch spaced banana plugs and the majority of high-quality, specialist audio cables. Transducers and crossover assemblies are neatly housed in a high-density, partical wrap cabinet, specially designed to minimize unwanted cabinet resonance, and high-frequency reflection. A little larger than the 6.5, at 15x10x10-1/2, the PBM 8 has a 8-inch poly cone mid bass transducer, utilizing an efficient long-throw, high-power voice coil. Low frequencies to 47 Hz are controlled by the rear-tuned port, with high frequencies beyond 20 kHz provided by a 1-inch ferro fluid cooled polymide dome tweeter.

Mfr.- Tannoy North America, Inc.

Price- $299.00 (PBM 6.5)
$448.00 (PBM 8)

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COMPACT AMPLIFIER

• PM-100 is a new single rack-space magnetic field power amplifier. This versatile power amplifier is designed for the requirements of broadcast and sound contractor use, as well as the demands of musical instrument and sound reinforcement applications. Among the unit's many features are a high-efficiency linear tracking, full complementary output stage, clipping eliminator circuitry, series/parallel mono capability, barrier strip inputs, front panel metering, and a stereo headphone jack.

Mfr. - Carver Corporation

Price - $629.00

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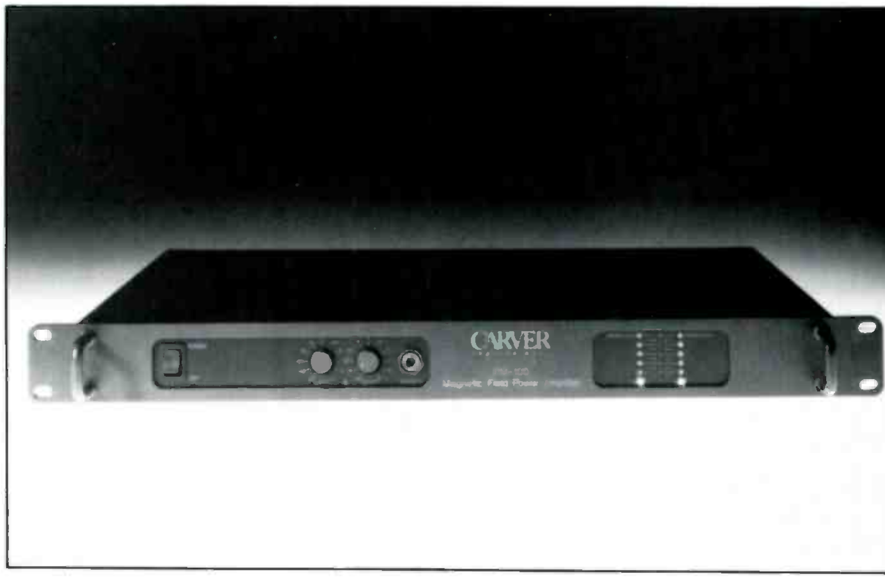

SAMPLED LIBRARY

• The EMAX Gold Series Library are specially prepared library disks provide high quality audio samples available for the EMAX product line. All sounds were originally sampled and processed using the Emulator III's true 16-bit linear circuitry and extensive digital processing functions. The sounds were then digitally transferred directly into the EMAX to preserve this sonic clarity. The result is an improved sample with increased fidelity and dynamic range. All Gold Series diskettes are provided exclusively on professional quality TDK media.

Mfr. - E-mu Systems, Inc.

Price - $15.95 per disk

Circle 65 on Reader Service Card

MIC SYSTEM

• Neumann's microphone series KM 100 is a modular condenser microphone system with a small diameter (21mm) capsule. The system consists of an active capsule that connects directly to the output stage forming an axially addressed microphone 3⅔-inches long. The active capsule, which is about 1.5-inches long and 0.8-inches in diameter, can also be connected to the output stage via a thin cable up to 150-feet long. The initial offering includes four interchangeable capsules: AK 30 omni, AK 40 cardioid, AK 45 cardioid with low-frequency roll-off, and AK 50 hypercardioid. A wide variety of accessories, including stands, extension tubes, goosenecks and cable hangers for mounting the mics and capsules, are immediately available. The fet 100's electronic circuitry is similar to the studio condenser microphone, the Neumann TLM 170. These hybrid, transformerless electronics result in a microphone which has 4 dB less self-noise, 3 dB higher output level and a 12 dB wider dynamic range (124 dB which is 28 dB greater than a 16-bit digital recorder) when compared to the KM 83/84/85 series.

Mfr. - Gotham Audio Corporation

Price - $600.00 (KM 140)
$725.00 (KM 150)

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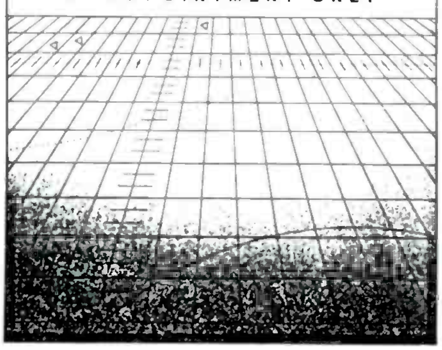

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People, Places... & Happenings

• **BASF Corporation Information Systems** has promoted **Terence O'Kelly** to the position of Director of National Sales for Audio/Video Professional Products. Mr. O'Kelly most recently held the position of National Sales Manager for Audio/Video Professional Products. He is credited with the development of the BASF Tape Clinic program for dealers and duplicators and has written a series of technical publications dubbed "The Inventor's Notebook."

• **Steven W. Terry** has been appointed Chief Operating Officer for **Mediatech West**, the three-year old offshoot of Chicago-based Mediatech, Inc., a commercial tape duplication/distribution organization. Mediatech West has embarked on a multi-million dollar expansion program to update its current 60,000 square-foot facility, and to augment its production capabilities with 15,000 square feet of additional space. In his new position, Terry will be responsible for all videotape post-production operations and implementing the company's expansion program.

• According to **M. Hosoda**, president of **Otari Electric Co. Inc.**, the company is integrating its offices with easy access to central Tokyo. In addition, in April, 1989, the company name will change to Otari, Inc. Otari celebrates its 25th anniversary in April.

• **John Johnson**, President and CEO of **DOD Electronics**, and three investors, **Charles Chewning, Tom Henderson**, and **Robert Henderson**, have purchased **DOD Electronics**, buying out the interests of **Mr. David O. DiFrancesco**, former partner with Mr. Johnson in the ownership of the company. The change of ownership brings new financial strength and experience to the operations of the company.

• **Orban Associates** announced the appointment of **David Roudebush** to the position of Marketing and Sales Manager for Orban's professional products (their Blue Panel line). Roudebush will be responsible for worldwide marketing and distribution of Orban's existing and upcoming products, excluding the OPTIMOD and broadcast transmission products which will continue to be managed by **Howard Mullinack**.

• **Russell Johnson**, chairman and founder of **Artec Consultants Inc.**, announces the recent addition of **Walfredo Toscanini** to Artec's team of acoustics and theater design specialists. Mr. Toscanini will be working on long-range planning, feasibility studies, and business development. After receiving a Masters degree in Architecture from Yale University, Mr. Toscanini did post-graduate work at the Wagner Festival at Bayreuth, Germany. Working with Thomas Munter, the internationally respected German theater architect, his research involved the study of elements of theater design and analysis of traditional and modern theater architecture. Mr. Toscanini had recently co-authored a photo-biography of his grandfather, Arturo.

• **Solid State Logic** announces the appointment of **Colin Pringle** as head of Marketing. He will be based at SSL's Oxford headquarters. He has worked in the UK, USA, and Europe, and brings with him detailed knowledge of audio and broadcast media worldwide.

• **Denon America, Inc.** President, **Mr. Takayasu Yoshida**, announced the promotions of four executives to new management positions in the company which is a subsidiary of Nippon Columbia, Japan. **Robert Heiblim** has been promoted to Executive Vice President. **Akira Saito** has been promoted to Senior Vice President. **John Langan** is now Vice President. **Ken Furst** has been promoted to Director of Marketing, a newly created position which reflects the expansion of Denon's marketing activities.

• **Berklee College's** Music Production and Engineering program asked sound designer **Jay Rose** to re-work their Production for Film and Video course. While some of the course topics are what might be expected (editing music against picture, or mixing for stereo TV), others range from production house salaries to client relations. Rose also includes a generous dose of film history. The course is part of a four-year degree program aimed at both musical and technical training as well as sensitivity to the needs of various media. According to Berklee's Department Chairman **David Moulton**, when the students leave the school, they will have been prepared with a full set of survival skills to adapt to a volatile industry.

• **New York Technical Support, Ltd.** in conjunction with **Al Theurer**, has recently formed a separate installation company. Theurer was previously the lead wire person for New York Tech. The new installation company is doing business as New York Tech and is offering site planning, computer-aided services, ergonomic optimization, panel design, installation and wiring design, and AutoCad operation. In addition, separate wiring services such as patch-bay assemblies, re-wiring and custom harness construction are available.

• **Georg Neumann GmbH** is celebrating its 60th year. In honor of the occasion, **Gotham Audio Corporation** dusted off an old photograph. Shown is **Georg Neumann**, his wife **Elie**, and **Jerry Graham**, at the 1967 AES Convention in New York City. Graham, Gotham's Eastern Sales Manager is currently celebrating his 25th year with the company.

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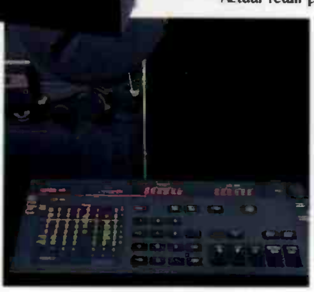

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