SOUND & COMMUNICATIONS

Volume 34 Number 10

October 1988

On the Boardwalk

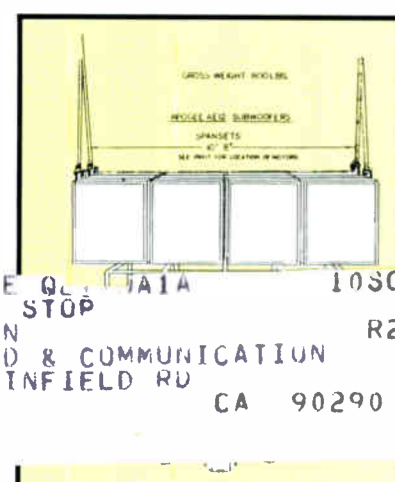
High rolling, high frequencies and other sound criteria coexist in the east coast gambling mecca of Atlantic City. The big-name shows keep the crowd gambling in the casino. And the sound systems keep the crowds coming. Sound and Communications investigates the leading casino sound systems along the boardwalk.

Sound and the Sports Arena

Concrete and steel fight speech intelligibility and music. The war was prevented in San Diego. How did they do it?

No-Feedback Sound

Temporary installations on a massive scale can cause headaches. There were few at the Omni Center because of good planning and equipment for the Democratic National Convention.



Previewing AES

This preview article is your guide. The theme is "A Century of Technology" and the workshops, papers and technical sessions promise an education.

Sound Reinforcement

Academics have their own d
processing — including a
modems to radar. Audio si
however, has its own definit
applications. Gary Davis a
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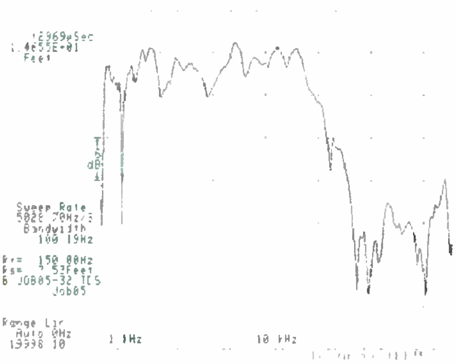

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SOUND COMMUNICATIONS

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EDITORIAL

A New Look

This month, we are proud to introduce a new look for an old friend—from cover-to-cover, we've given *Sound & Communications* a completely new design. Not a radical departure, but a design that is sensible and dynamic, serious yet inviting, with a clear and consistent look for the departments, columns, and features.

While our outward appearance has changed, our editorial commitment has not. For over thirty years we have been addressing the interests of professional contractors, designers, consultants, engineers, and system managers involved in the areas of sound reinforcement, paging and intercommunications, background and foreground music, alarm and security, and teleconferencing. We will continue to address those interests: our new look is intended to enrich the magazine, not to alter its direction or purpose.

In the last thirty years, new advancements in technology have brought much change and growth to this industry. New developments in the areas of fiber-optics, digital signal processing, and satellite transmission have brought new applications for these technologies in the areas of security surveillance, sound reinforcement, and business music. As an industry, we are obligated to grow with and adapt to these new developments. But growth and adaptation can only come with knowledge and awareness. As a magazine, *Sound & Communications* is dedicated to bringing you all the details of the latest advancements, the pioneering technologies, and the newest applications, helping you to stay informed and aware.

We'll continue to bring you thought-provoking and informative feature articles—lab test reports, installation profiles, business management techniques, market analysis, technical tutorials—and we'll continue to bring you our exclusive special issues: our annual surveys of the contracting market and the state of the industry's economy, as well as our annual *Blue Book* directory of manufacturers, products, and manufacturer's reps.

But there are some surprises in store as well: new authors and contributors, and new columns on management, finance, and technology. We intend to be as responsive as possible to your changing needs—but we need your input. Let us know what you think about the new design. Let us know in what subject areas you'd like us to expand, what business management topics you'd like us to examine, what new technologies and applications you'd like us to explore.

None of the changes we've made is carved in granite. We expect a period of transition during which we'll be testing and fine-tuning our new design—just as you would any new "system." Your comments are going to be an important part of that process.

Bill Intemann

Bill Intemann
Managing Editor

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NSCA-TV News • NAMM-TV News • CES-TV News
The Music & Sound Retailer
Home Entertainment

NEWSLETTER

***EXCLUSIVE:* ELECTRO-VOICE REORGANIZES**

Electro-Voice has reorganized its operations following its acquisition of Cetec. Paul McGuire takes on more of the overall responsibilities of Electro-Voice. Bob Pabst, president, takes on more of the responsibilities for corporate direction of all Mark IV audio companies. At press time, full details were unavailable; however, increased responsibilities for current staff were anticipated.

FANE APPOINTS TAYLOR DIRECTOR OF U.S. OPERATIONS

Fane Acoustics, based in England, has appointed Terry Taylor as Director of U.S. Operations. Taylor's plans for Fane include making more of an impact in the U.S. market. The U.S. operations will be based in Chicago. Taylor previously had been head of sales and marketing at Toa and had been part of the advertising and marketing team at Peavey in Meridian, MS. Fane will be introducing their new Court Line of speaker enclosures at AES.

TRIDENT BUYOUT

Trident Audio Developments Ltd. has been sold to Relyon Group PLC, a publicly traded British company. After more than 16 years as directors of Trident Audio Developments Ltd., Malcolm Toft and Jack Hartfield have sold their entire interest in the company. Toft and Hartfield will remain as active consultants to Relyon. Bud Brimberg has been selected as managing director. Brimberg, an American-trained attorney, has two successful start-up companies to his credit: ProCreations Publishing Company and Plan-A-Flex Designer Systems. Wayne Freeman will be Director of International Sales and Marketing for Trident (UK). Freeman has been part of Trident's U.S. distribution team for several years, having joined Lon LeMaster as part of Trident Audio USA in 1986.

NSCA FORUM ADDRESSES CONCERNS

At a forum held by the National Sound & Communications Association (NSCA) a new Industry Advisory Conference (IAC) of Underwriters Laboratories had been decided to be formed due to the lack of quality communication between the commercial/pro audio industry and organizations such as the Underwriters Laboratories and the Canadian Standards Association. The panel of ten to fifteen industry representatives will be chosen from a larger working committee of primarily — but not exclusively — NSCA member companies, known as the Product Safety Group of NSCA.

MDA EXPANDS INTO SOUND AND COMMUNICATIONS INDUSTRY

The Music Distributors Association (MDA) has announced a new program for membership expansion that will encompass members from the sound, communications and electronic industry. MDA was organized in 1939, originally as a music distribution association. The membership has diversified to include manufacturers, importers and exporters covering all facets of the musical instrument and pro-audio industry. MDA, currently made up of approximately 90 companies, is the largest and most active of all the trade associations in the music business other than NAMM.

RING COMMUNICATIONS ESTABLISHED

Ring Communications Inc. has been established as a privately-held communications company. Peter McLean, former Ring Group vice president, is now President and Chief Operating Officer of Ring Communications Inc. Craig Krsanac and Kjell (Ken) Salem will join him as officers and part owners of the new company. Ring Communications Inc. had been established following U.S. corporate restructuring by Scanvest Ring A/S.

The new firm will specialize in the marketing and installation of office, school and industrial intercoms and other specialized communications systems. Ring Communications will also offer parts and honor commitments for hardware, software and warranty support to customers of Ring's Enhanced 9-1-1, automatic call distributor and intercom systems.

NEWSLETTER

AES CONVENTION IN HAMBURG

The 86th AES Convention will be held in Hamburg on March 7-10, 1989. More than 5000 participants from all parts of the world are expected to discuss new knowledge and modern application techniques in audio technology.

APOGEE SOUND EXPANDS INTERNATIONAL DISTRIBUTION

Apogee Sound, Inc. has expanded their international distribution network to include seven new distributors throughout Europe and in Australia to handle an increase in export demand. The new appointments include: Entertainment Services Australia, Australia; Titan Audio, Belgium; Paul Farrah Sound, England; Orthophono Professionale Audio, Italy; Focus Showequipment BV, Netherlands; Sony Espana, Spain and Audio Rents B-AG, Switzerland.

KLARK-TEKNIK DISTRIBUTOR FOR MILAB MICS

Klark-Teknik Electronics has been named the exclusive U.S. distributor for MILAB Microphones. A few of the microphones presently available are the VIP-50 multi-pattern large diaphragm studio condenser, the DC96B large diaphragm cardioid studio condenser and the LC28 cardioid condenser live vocal/studio microphone with HP filters and attenuation pads.

SYN-AUD-CON WORKSHOP

Synergetic Audio Concepts (Syn-Aud-Con) of Norman, IN will sponsor a workshop on Concert Sound Reinforcement on January 17-19, 1989. Don and Carolyn Davis will host the workshop. Workshop Chairman, Will Parry of Maryland Sound has chosen his staff from Showco, Electrotec, and Audio Analyst: Clay Powers, Mick Whelan, and Albert Leccese. David Scheirman of Concert Sound Consultants will be the facilities coordinator for Will Parry.

GE AMERICOM WINS ALSCOM SATELLITE

GE Americom Communications, Inc., has signed a contract with Alascom, Inc., the long-distance carrier for Alaska, to provide a successor satellite to Alascom's Aurora I, together with tracking, telemetry, control and in-orbit protection services throughout the 12-year design and fuel life of the new spacecraft. GE Americom is to own up to 10 transponders on the new spacecraft and to launch a second satellite to provide in-orbit protection for Alascom, according to the contract. The protection satellite assures service restoration capability for all C-band spacecraft in the GE Americom fleet. The new shared satellite, designated C-5 by GE Americom, is scheduled for launch in May 1991, and is being designed and constructed by GE Astro Space Division.

JERRY RAYMOND ICIA CHAIRMAN RETIRES

Jerry Raymond, Chairman of the ICIA Board of Governers and President of the V.I.P. Division of Willoughby's in New York City, has retired after 42 years. Raymond has been an active member of ICIA for more than 20 years, serving on numerous committees as well as a regional Board member which culminated in his being elected as chairman in 1985. The ICIA contact at Willoughby's is now Ms. J. Christine.

ENTERTAINMENT EQUIPMENT TAKES OVER THEATRE EQUIPMENT AND SERVICE

Entertainment Equipment Corporation, a newly formed organization involved in the sale and distribution of entertainment related equipment, has taken over the Buffalo operations of Pittsburgh-based Theatre Equipment and Service Company. Entertainment Equipment sells products ranging from public area seating, and complex sound and lighting systems, to state-of-the-art audio/visual equipment used by businesses and educational institutions. Recently the company set up a 24 hour hotline service to assist customers in emergency situations. "The new around-the-clock emergency service coupled with the purchase of Theatre Equipment's parts and supply inventory will enable us to address customer operating problems very quickly," stated James A. Lavorato, company president.

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The SPL War

by Gary D. Davis

Today, sound systems capable of producing higher SPL with lower distortion and wider bandwidth than could be found in the finest theaters back in 1968 are now routinely carted around by small bands. Contractors have taken pride in the perfection of their art, technology, and science, and most of the people who manufacture and sell sound equipment are pleased about the quality of sound which can be delivered for reasonable dollars. Yet somewhere along the line we just may have allowed the object of all this technology to elude us.

Sound reinforcement systems for entertainment applications were initially designed to amplify the music and vocals, to make both clearly audible to those assembled, with reasonable faithfulness to the original, unamplified sound. We all know that headroom is one of the keys to realistic, low-distortion reproduction, and that is why systems having tens of multi-hundred watt amplifiers driving high-sensitivity speakers are now commonplace. Excess power provides a margin of protection from distortion, or at least it is supposed to.

Lets consider a hypothetical example: a simple, mono speaker cluster that needs about 100 watts to punch through a modest sized night club audience with enough level to be heard clearly. In the "good old days," the contractor or system designer might have installed a 200 watt amplifier, providing 3 dB of headroom over the loudest levels that might be used, and 6 to 8 dB above typical levels. (For paging, 3 dB would still be considered sufficient.)

Ah, but along came the Beatles, and we learned that a few speakers with a few small amps were not enough to cover the thousands at Shea Stadium. The Grateful Dead later filled a stage from one end to the other with speaker stacks over 10 feet high; they knew how to use headroom to good advantage, but most others didn't get the point. People (soundmen in particular) saw "more" as being "better." One rock festival outdid the next with humongastacks, and we were fully involved in the undeclared SPL War.

Well, thinking back, maybe 3 dB was really not sufficient, and so the club owner eventually had reason to install a 1 kW amplifier stack that provided 10 dB of headroom. That should have done the trick. And indeed it would have, if the nominal level had remained at 100 watts. Unfortunately, that is not what happens in many cases.

In many instances, the band's sound man, or "soundwoman" (we're not sexist—for that matter, anyone who controls a sound system), has grown up learning to judge how *loud* the sound is by evaluating how *distorted* it is. These people actually believe that there is no "guts" in the sound system unless every meter's pre-clip peak LED is glowing steadily, and every amplifier's "clipping" LED is lit up. The system with 3dB of headroom was operated with virtually no headroom: the unsophisticated operator would simply "turn it up" until the distortion became very audible. When the owner "fixes" things by installing a sound system with 10 dB of headroom instead of 3 dB, these operators do not take advantage of the extra 7 dB margin of protection from clipping and other grunge—they just turn up the level another 7 dB (or more) to get that distortion right where it ought to be.

You might think the perpetrators of such outrageous behavior would be discouraged by the sheer acoustic energy bombarding their ears; after all, the SPL was probably hovering around an almost bearable 115 dB on stage with the old 200 watt amp running at the rails. But now, with the 1 kW system running at 1000 watts (forget the 10 dB headroom), the level rises to a painful 122 dB SPL! (Assuming the speakers don't give up.) Why don't they turn it down?

Well, for one thing, there is the issue of hearing loss. The band has been playing music loudly several times a week—maybe several times per day. They, and especially their soundman (who is located in front of the main speaker stacks), are bound to experience a certain amount of hearing loss. This is known as "permanent threshold shift" and sufferers simply do not perceive the music to be as loud as it seems to the audience. Then, too, the band and soundman experience an additional "temporary threshold shift" due to immediate proximity to stage monitors, house speakers, guitar amp speakers, and so forth. This additional threshold shift further deafens them so that, around the time that the first wave of disgusted patrons (or guests at a private event) march forward with a small white flag and ask them to "please turn it down a little," the soundman has probably decided it needs to be cranked up a few dB (eliminating what slim headroom had remained). This judgement is seldom moderated by the band—even if they do care about the sound in the house,

Davis is president of Gary Davis & Associates, Topanga, CA, and is the author (along with Ralph Jones) of the Yamaha Sound Reinforcement Handbook.

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The tremendous success of the Tannoy PBM series of reference monitors is by no means coincidental. Since the introduction of the world renowned NFM-8 nearfield monitor, much time and effort has been spent on discerning the needs of the mixing engineer and the applied requirements of "playback monitors". The PBM Line exemplifies this commitment to excellence in reference studio monitoring. These compact loudspeakers sport robust poly cone mid-bass transducers utilizing efficient long-throw, high power voice coils. The low frequencies are carefully controlled by optimumly tuned ports located on the rear of the loudspeakers. Hi frequencies are provided by Hi Power ferro fluid cooled polyamide dome tweeters which extend H.F. bandwidth beyond 20KHZ. 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 3/4" spaced banana plugs and the majority of high quality, specialist audio cables. Transducers and crossover assemblies are neatly housed in a stylish, high density, partical wrap cabinet, specially designed to minimize unwanted cabinet resonance, and high frequency reflection. In summarizing, we have left the best feature of all for last "price versus performance."

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CONSULTANT'S COMMENTS

they can't hear it much because they have their own, separately adjusted monitor speakers.

As a matter of fact, in small clubs the stage monitors may contribute significantly to the level in the house. This can be a problem when one musician figures he is more important than another, so his monitor should be louder. The next one then signals the engineer at the monitor board to crank up his level, and so it goes. Damn the torpedoes, full speed ahead!

Actually, the "please turn it down a little" request was probably a polite statement of the peace envoy's true feelings, which were really something like this: "I am in great pain, I can barely hear my date when she's shouting in my ear, I'm getting hoarse, and if you don't have the decency to turn down the sound system, we're out of here, buddy! The reply from the band (or the soundman) might be, "#&$*@!" (loosely translated, "buzz off, we know what we're doing,"). More civilized soundmen may make a weak attempt to keep the peace with a comment like, "We've been playing dates like this for years, and we know that people need to hear it loud so they can dance. We can turn it down a little, but the band wouldn't like that. Say, why don't you just move further back?"

Well, the truth is, these folks are not about to turn down the level willingly, and the person who hired the band is probably too busy with other things to deal with the issue. What is worse, a lot of ego is involved, and that's just plain difficult to deal with for anyone who might be so bold as to request a more reasonable SPL.

If you are not already aware of it, here's the most significant issue: SPL alone does not determine the perception of loudness. Distortion is the other primary factor. What's worse, distorted sound at a given high SPL is much more damaging to the hearing than undistorted sound. Many of us can handle 130 dB SPL on momentary peaks without great discomfort, yet most of us will scream with displeasure at 115 dB SPL with 50 percent distortion. While an SPL meter is a useful tool for reference, the bottom line is that it won't tell you what you really need to know. That is, it's too loud when a significant number of guests/audience think it's too loud.

Think about it. How many times have you, personally, felt like leaving an event prematurely simply because the sound was beating you up? This author has had to do so far too often. It is one thing if people come to an event specifically to hear the music. Then it should be loud; perhaps we should be willing to accept 3 to 6 dB more than we might otherwise find comfortable, but no more.

When the event is not a concert, then the music is an enhancement to the other activities, and should not intrude on every conversation. It should not drive people reeling away, holding hands over ears and/or wincing. It seems as though the bands have been attacking us more and more (and with some very heavy artillery.) Enough!

The sad fact is that far too many club dates, parties, and other events that should be joyous are marred by assaulting sound that ranges from uncomfortable to outright painful. It is very difficult for the people being injured to get out of the line of fire, and even more difficult to get their point across without causing very bad feelings on the part of the performers and/or their crew.

Consider the following:

1. If even one person comes up to ask that the level be turned down, pay attention (that person may have been urged to do so by several guests).

2. If several people come up separately, you can assume it really is

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Teleconferencing Acoustics: Echo Cancellation Technology

by Steven J. Orfield

The use of teleconferencing, whether video or audio, has posed some very difficult problems within the field of audio design, and foremost among these problems is the issue of conference intelligibility. While it would seem that the need for intelligibility is obvious within this context, it is apparent that many audio designers neither develop intelligibility standards for these systems, nor determine a calibration procedure for use between teleconferencing sites to determine whether the systems are at minimum intelligibility levels. As with so many problems in audio, it is often assumed that the purchase of high-quality equipment will resolve most design problems.

Intelligibility theory is somewhat complex, and a familiarity with its basic tenets is fundamental to teleconferencing design [see Orfield, S. "Speech Intelligibility," *Sound & Communications* (October & November 1986)] As with most problems in acoustics, teleconferencing intelligibility is generally considered via three possible paths of evaluation:

1. Source room and system.
2. Transmission path and system.
3. Receiver room and system.

Problems with intelligibility in this

Orfield is president of Orfield Associates, Minneapolis, MN.

setting are generally due to one of three variables:

1. Signal level (S/N).
2. Frequency Distortion.
3. Time Distortion.

Thus, it is necessary to determine minimum signal-to-noise ratios, maximum permissible background noise levels and minimum frequency response characteristics needed. After this has been specified and accomplished, another factor often becomes painfully obvious: the presence of transmission-related echoes.

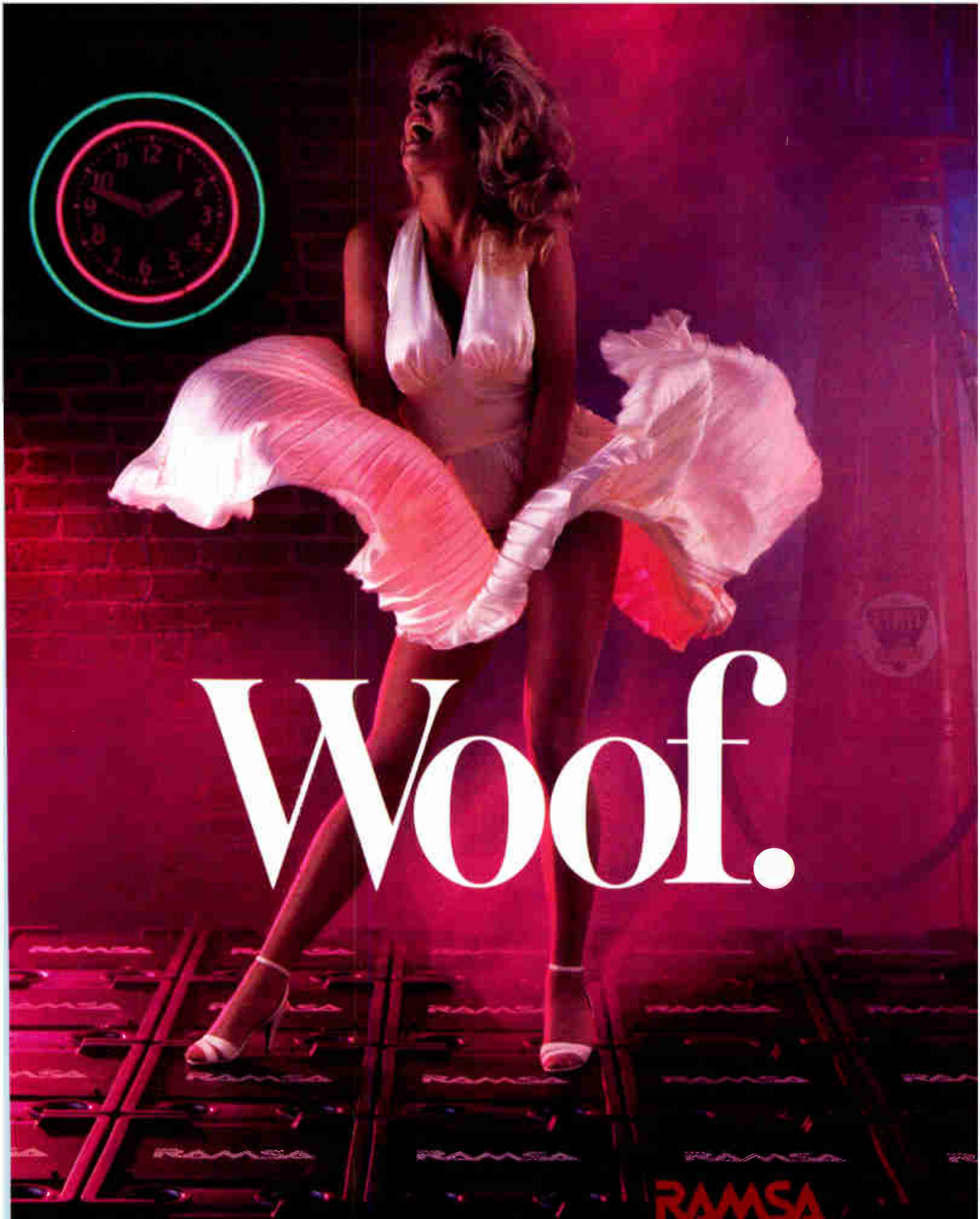
The basis of echo problems is partially the delay caused by transmission time offsets from satellites and phone systems, and partially the effects of room response on the signal received at the local site. These problems are often dealt with by a device known as an "echo canceller", and this device attempts (within the limit of its potential adjustment) to respond to these transmission anomalies.

Generally, echo cancellation devices attempt to assess the secondary time response which follows the voice communication. This may be the echo produced when the signal received in the local room reflects from surfaces within that room, reenters the local microphone system and is reheard within the sending room after a short delay. It may also be the signal heard via direct transmission from the local loudspeaker to the local microphone within the receiving room, thus "short-circuiting" the system and being sent back to the remote talker after a time delay. In either case, this audible echo creates both quality and intelligibility problems for the user, as echoes mask and distort the speech signal.

The audio industry has responded to this problem with echo-reducing devices. There are currently a number of manufacturers of these types of devices, and the products which they produce are either independent (i.e., the NEC Acoustic Echo Canceller) or they are part of a microphone system (i.e., the Shure Teleconferencing microphone system.)

Since the audio field is in the early stages of the use of these devices, the manufacturers have not generally documented either the performance of the system or the needed performance of the room to be compatible with the system (some specific tests of systems will be documented in a later article concerning the function of these systems).

(continued on page 74)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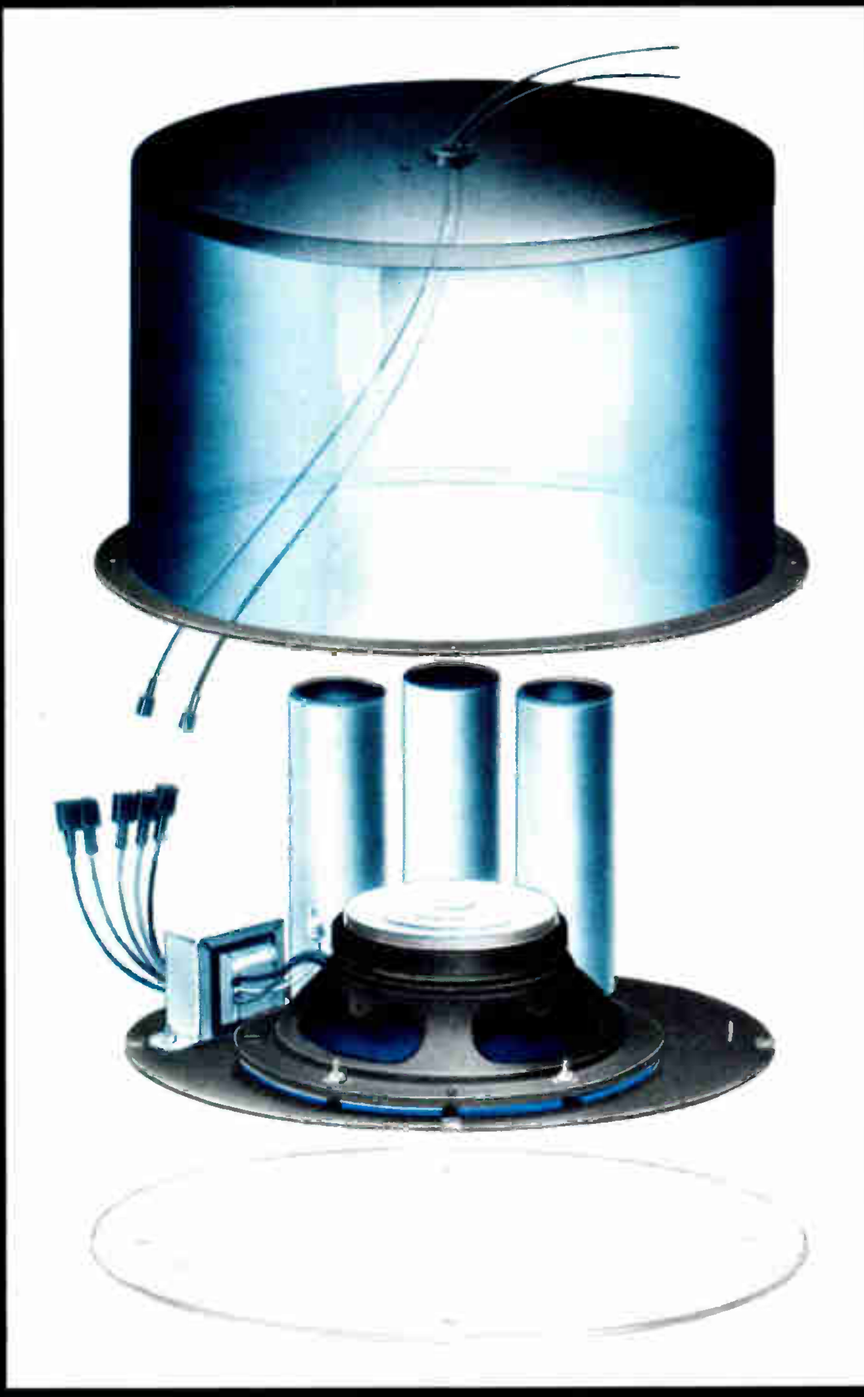

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Circle 207 on Reader Response Card

The 1988 Democratic National Convention

by John Krementz

This past summer the media was saturated with reports from the home of the Atlanta Hawks—the 20,000 seat Omni Center. Well, it wasn't a Hawks game, and it wasn't a rock concert: it was the 1988 Democratic National Convention. For a week on national television, we were bombarded with the slick productions aired each day from Atlanta—all without a single squeal of feedback. An impressive accomplishment: due to the massive structures built for the network studios and secret service, the complex staging requirements, various camera locations, a small army of betacam crews (not to mention the large number of delegates in attendance), the incredibly complex intercommunicatons requirements for all of the different groups assembled—sound reinforcement was a complicated task, to say the least.

The audio contract was made directly with the Democratic National Convention Committee, an organization with limited "show biz" experience. Fortunately, sound was considered to be of paramount importance, and the committee hired Burns Audio of Sun Valley, CA, early in the game. Due to the scope of the project (and the number of subcontractors), Burns Audio had all 11 audio drawings included in the master construction plans. The Ted Turner Construction Company handled the principal $3 million construction modification contract. The electrical contractor installed all of the audio power runs, and the rigging contractor installed the loudspeaker cluster hoist-motors and hanging pick-points in advance. Since the audio plans were drawn up first (and included in the master plan), the subsequent lighting plot took a back seat and had to be worked around the speaker clusters without blocking their coverage.

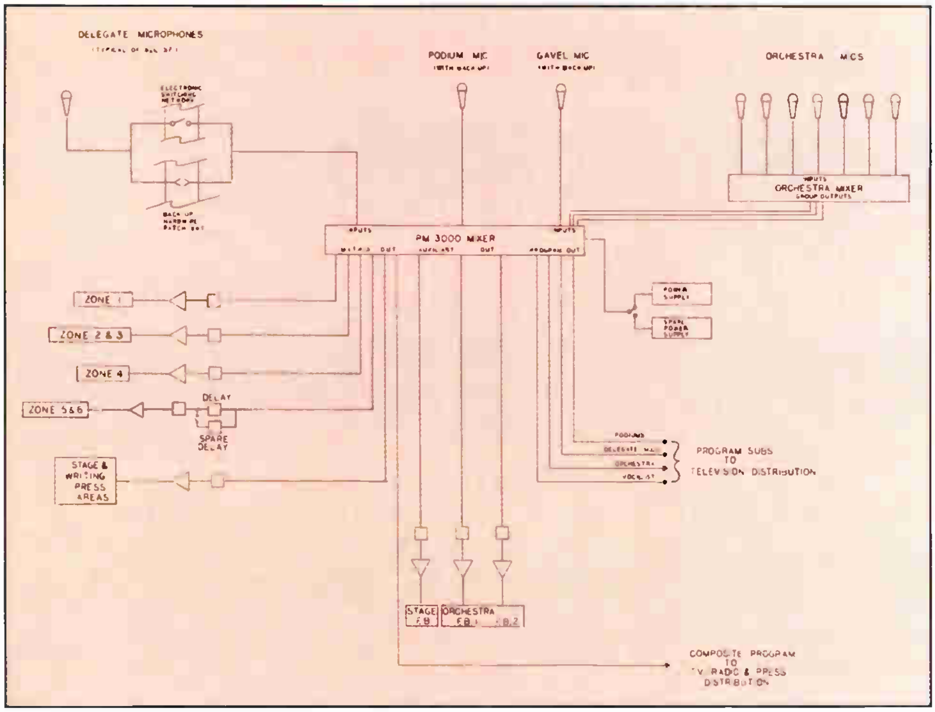
Unusual seating arrangements were created by the combined requirements of producing a national television broadcast and staging a live mega-event (including dual large-screen video projection). These seating arrangements proved to be a blessing in disguise: the room was broken up into a number of smaller areas and, acoustically, the reduced volume and multi-faceted surfaces modified the room-modes, decreasing reverberation and the degree of energy contributed to the reverberant field, as well as improving loudspeaker array coverage angles. However, two areas were found to have excessive reflected energy. One was a writing press area backstage behind the wide screens, and the other was a small section along side the glass network towers. These problem areas were corrected with Apogee AE-3, and AE-1 speakers on a delay—effectively raising the direct level above he reverberant level.

The stage is set: clusters, lighting trusses, and network "skybooth."

The loudspeaker clusters were divided into three main sections: a trio of main-arrays, with 15 Apogee AE-5s; one delayed array, with 17 Apogee 3x3s; and a subwoofer section, with two sections of four Apogee AE-12s. The 3x3s are a three-way fully horn-loaded system comprised of two 15s, a two-inch midrange driver, and an array of four piezo tweeters—these loudspeakers maintain 60 x 40 degree pattern control even at low frequencies. The AE-5 2-way trapezoidal shaped cabinet with a 12-inch woofer, and a one-inch 90 x 60 constant directivity horn. The AE-12s are a double-18-inch vented enclosure with a flat response to 32 Hz. These systems use dedicated processors which include crossover, time alignment delay, EQ, and RMS limiting functions.

The sound system block diagram shows all inputs and outputs. The podium was covered by the "Presidential" Triple-Shure SM-57. Two of the 57's are processed by a dbx 160X "over-easy" limiter, and a Klark-Teknik DN405 Parametric EQ. The third 57 was a secondary emergency stand-by which is hardwired to a Shure M267 mixer that could be patched at the amp-racks in case of a catastrophic front-of-the-house equipment failure. Using custom active press bridges with transformer-isolated outputs, the house PA mix provided audio distribution on the floor for the smaller remote crews, as well as mobile radio crews. There was a similar press distribution network for the press overflow in the World Congress Center across the street. The audio and video feeds for this location were sent via fiber optics channels provided by AT&T. Five Yamaha M406s were used as sub-mixers to a Yamaha PM 3000-40, output to

Krementz is an acoustical consultant in Philadelphia, PA.



Block diagram of sound system inputs and outputs.

Klark-Teknik DN360 1/3-octave house EQ's and a dbx RTA-1. The dbx analyzer uses the program input as a reference, and a microphone located in a typical audience location as a comparison, displaying the difference in one-third octave bands. All of the system power—55,000 watts—was provided by Crest 2501As, 4001s, and 8000s.

The clusters were hung on motorized hoists using a rigging chassis which was a hybrid between a standard "antenna-mast" type truss and custom attachment fixtures. The speaker hanging brackets (built by Burns Audio, designed by Burns Audio mechanical engineer Robert Brogden) were fabricated from aluminum and allowed total adjustment of both pitch and yaw. D-rings and Span-Sets were used to attach the speakers to the custom brackets, this allowed the units to array tightly adding to their cohesiveness.

Delegate Rostrums

Each of the 50 states and six US territories had a rostrum with a Shure SM-57 microphone, an AT&T personal computer terminal, and a telephone. The AT&T terminal was used for voting, as well as sending and receiving messages, while the mainframe was located in a dedicated room backstage. The majority of these rostrums were located on the main floor, actually a false floor raised eight inches to allow for the concealment of all wiring; the remainder of the rostrums were distributed in the first tier of seating. In all, over 12,000 feet of Belden 9451 cable was installed for these microphones, with the understanding that it was expendable: the schedule to strike the sound, video, camera, and computer systems was 12:00 noon following the end of the convention at midnight. Crews worked through the night to prepare for the arrival of massive construction cranes, scheduled to roll in at noon to remove the network towers and skybooths. These structures were then to be trucked to New Orleans for—you guessed it—the Republican National Convention.

Somewhere in convention history, the concept of a dedicated switching panel to lock out all delegate microphones except the state currently recognized by the chairperson became necessary, for security reasons. This must have been prompted by a particularly unruly convention involving sabotage between states. Even as recently as four years ago, each of the delegate microphones were bolted to a welded stanc which in turn was bolted to the false floor. The mic cable ra through armored BX-type cable directly into the welded stand- leaving no exposed wire.

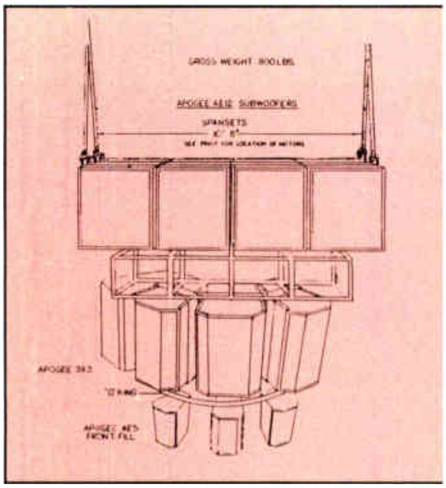


Diagram of cluster assembly.

This year Gene Richards of New York, NY, a free-lance audi engineer, designed and built a delegate microphone switcher. The system assigns one of the microphones to a pair of RTS preamps which send a line-level signal to the consoles. A parallel hard-wired patch bay serves as a backup. Four audio assistants wearing Motorola HT-600 walkie talkies fitted with custom headsets were positioned on the floor with Cetec Vega RF microphones in the event of a failure (in fact, one did occur when the Vermont delegate microphone cable got pinched between two bleachers and the RF was called into service).

RF Madness

The RF situation was incredibly complex and elaborate: the authorized list of frequencies ran 22 pages, covering from 25 mHz to 39975 mHz! Burns Audio had already been assigned four frequencies in the the 169-219 mHz range. Unfortunately, there were 192 other assignments in that same VHF band. Three of their four frequencies had other transmitters within 100 kHz rendering them unusable. The only solution was to use the IFR RF spectrum analyzer each day to search for "windows." Burns Audio technicians were able to find three additional slots and units were shipped from L.A. on those frequencies.

Conditions were so intense that Doug Miller and Royce Leonardson of the Atlanta FCC were enlisted to act as deputies and monitor violators. Stationed at various places they would use a highly directional antenna (called a back spiral cavity) to locate unauthorized transmitters within the building. The FCC officials were kept busy throughout the convention by the sheer number of roving Betacam units, many of whom arrived without coordinating their radio equipment.

> "Productions aired each day, all without a single squeal of Feedback. An impressive accomplishment. . ."

This installation was unusual in that the completion dates were absolutely firm and the work schedule was almost "paramilitary." This project had all the eyes and ears of the world watching and listening intently: this exposure, combined with the execution of the installation and the system's flawless performance, shines as an example of what this industry can do. ■

Diagrams courtesy of Burns Audio.

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Circle 211 on Reader Response Card

Showroom Sound Reinforcement

by David Starobin

Last month, we took a look at the sound systems currently being used in the lounges and night clubs of Atlantic City. This month, we take you on a tour of the casino showrooms—the "crown jewels" of Atlantic City.

Everyone loves a show, and there's something for everyone in Atlantic City. On any given evening here, you can choose from a dazzling assortment of shows: a burlesque revue, a Broadway play, a laser/electronic music extravaganza, a world title boxing match, a Fifties revue, The Newlywed Game taping for TV, a March Of Dimes telethon, Morton Downey broadcasting live, magicians, jugglers, acrobats, trained chimps—you name it, there's a showroom here that's got it.

However, the real business of Atlantic City is gambling, not producing shows. With all the "comps" being handed out, ticket revenue does not begin to recover the cost of putting on these lavish spectacles. The shows exist for one thing, and for one thing only: to maintain the flow of people coming through the casino doors. The entertainment departments operate at a loss, and each casino sets its own priorities as to what type of shows it will produce, and how much of its budget will be allocated for those productions. These management decisions dictate to a large extent the type of sound system each casino will install.

The Sands Copa Room

The Copa Room has a sound system designed by Al Siniscal and Ken Newman, installed by Ken Newman and Dave Wink in August 1980. A-1 Audio of Los Angeles supplied the road-style system, complete with rigging. This was a slightly different approach from other installations in town: basically a touring system configured to the room, which the house liked and was able to install quickly.

The Copa Room hosts a wide variety of functions, from boxing, to revues and headliners such as Harry Belafonte and Lou Rawls. The system adapts to all of their varied requirements with ease. "A very important part of the flexibility and adaptability of the Copa Room sound system is the snake and cabling system," notes Bruce Raykiewicz, the Copa Room's lead tech.

The patch bay is the termination point for all of the connections to the equipment and cabling, and is another important factor contributing to the system's adaptability. The main mic snake, custom-made by Wireworks, is a parallel split that runs from the patch bay in the sound booth to stage left, stage right and upstage center. It's all in conduit and, says Bruce, "the parallel split is okay as long as at least one of the consoles has transformers across its inputs." 19-line boxes are used on stage to direct signals to the three stage points via multipin connectors. To provide for line level returns, there are 15 lines, terminating at the power amp rack on stage left and the patch bay in the sound booth. There is also a four line cross-stage snake. Additional lines to the patch bay in the sound booth are run from floor pockets along the stage apron and from the catwalk in the ceiling.

The system components include: six JBL/A-1 Audio 4350 four-way loudspeaker systems; two Myers 650R2 sub-woofers; JBL 4602 and Yamaha 2115 monitor wedges; 11 BGW750B power amplifiers; DBX, Orban, and UREI units for outboard limiting and equalization; JBL 4602 and Yamaha 2115 wedges; and an assortment of over 100 mics (including 9 wireless mics by NADY). The mixing console is a Yamaha PM2000 that dates from 1979 (Bruce's face lights up with mention of a new PM3000—the new console would allow the old one to become the stage monitor board). The Copa Room also provides direct boxes, a Baldwin grand piano, a Prophet synthesizer, a Rhodes piano, and guitar/bass amps.

The speaker system was designed for acts like Carol Channing and Bert Parks. There isn't enough SPL for today's high energy shows like the Rascals or Michael McDonald. What it lacks in loudness is made up for by the full, even acoustics of the room. The bass problem is adequately overcome by the pair of thunderous Myers 650R2's. Each of the six JBL/A-1 Audio 4350 four-way loudspeaker systems is bi-amped, using the BGW750B amplifiers. Four of the speaker systems are in a central cluster configuration above the center of the proscenium opening, while the other two systems are on opposite proscenium walls. The side stage monitors have been custom made by A-1 Audio. They are three-way systems powered also by BGW750B amplifiers. Additional stage monitor speakers are available, with additional BGW750B amps.

After the late show on Saturday night, magic happens in the Sands Copa Room. Within an hour the technicians have the stage struck while the hotel team clears the tables and chairs from the front half of the house. A huge lighting truss is lowered, a dance floor is put down, the doors open, and in come hundreds of revelers to dance the night away.

TropWorld Casino & Entertainment Resort

The Tropicana boasts the second largest stage on the east coast (Radio City Music Hall has them beat). The room seats

1700 patrons at long tables, with plush, cushioned banquettes where the Maitre d' seats his select clientele. Right now it is certainly the largest casino showroom in the world. Once the construction was completed, an additional $350,000 was spent on the staging, lighting, and audio equipment. The system features BGW750B and Crown Delta Omega power amplifiers, Yamaha's PM3000 console, and the Yamaha REV-1 digital reverb. The system's speaker cluster includes TAD 4001's, Community Light and Sound M4's, and JBL 2240's.

The Copa Room at the Sands Hotel, showing central speaker cluster and side-fill monitors.

This is, in many respects, a functional and versatile system. Signal routing and patching is very diverse and the room can handle just about anything. The in-house stage monitor system rivals many of its competitors' house systems. The speaker cluster, flying fifty feet above the audience, contains twenty-one of the Community Light and Sound M4 drivers as its mid-range components. The high, mid, and low end are powered by a roomful of the BGW and Crown amps: over 30 kilowatts worth. However, the room is severely lacking in bottom end. There are only two pairs of 18-inch sub-woofers, suspended in the air by chains: they cannot compete with the 21 M4's. (The 17,000-seat Spectrum Arena in Philadelphia has only 13 M4 drivers—and they are plenty loud!)

The Trop has other problems. When construction was completed, the discovery was made (on opening night) that the room was, basically, acoustically unintelligible: people talking on stage could not be understood. A consultant was called in to alter the room's acoustics, and another $120,000 was spent on acoustic treatment on the side walls. This fiberglass batting is good for controlling reflections down to only, say, 120 cycles. That, coupled with the live surfaces on the huge stage, make what little bass there is just rumble around forever, just as unintelligible as the upper ranges were before they were treated.

Other factors contributing to the difficult acoustics are the hard ceiling (parallel to a floor full of hard table tops), and the 100 feet of glass fronting the lighting booth high on the back wall. Unfortunately, the glass is set at the perfect angle to reflect the sound from the clusters right back to the stage. Performers here often ask the soundman to remove the apparent delay from the speakers when, in fact, the culprit is those reflections.

The room definitely has the potential to sound great: Maryland Sound did some amazing things with off-stage bass bins for Patti LaBelle and Frankie Valli. I sincerely hope that the Tropicana can correct those flaws that prevent this room from becoming the gem that it truly can be.

Trump Plaza

Trump Plaza has recently completed a new installation in their showroom. The design specification and installation was done completely "in-house" by Jay Paul and his crew, with the assistance of one of Atlantic City's only TEF analyzers. Paul coordinated with TekCom Corporation, Eastern Acoustic Works, and Crown International.

The new system includes 10 EAW KF850 full-range loudspeaker cabinets, 6 Meyers USW sub-woofers, 10 Crown MA600 and 3 Crown MA1200 amplifiers, the Yamaha PM3000 mixing console, PAS coax wedges for onstage monitoring, and a Wireworks mic cabling system (with a 50-input stage box featuring a three-way split using Jensen transformers). A major audio/video tie line system was installed at the same time, featuring two video cameras, four video monitor tie lines, and six intercom lines.

The Claridge

The Claridge is the smallest casino in Atlantic City, and what sets its showroom, The Palace Theater, apart from all the others in town is its policy of presenting only Broadway shows. Over the years they have presented cabaret versions of *Sophisticated Ladies* (with the Mercer Ellington Orchestra), *Irma La Douce*, *Woman of the Year* (with Elke Sommers), and *42nd Street* (with Peter Marshall).

I recently attended *Applause*, starring Phyllis McGuire, and can verify that Phil Hartshorn has adapted this room to present shows in a pleasing and enjoyable manner. . . and that's not easy, given the kinds of shows featured here, and the nature of the room itself.

The room is small in comparison to the other showrooms (it seats only 600 people). The stage is thrust ten feet into the audience, and the overhead speaker cluster is also ten feet beyond the proscenium line, "almost an arm's length" from the first lighting catwalk in the ceiling. "A lot of sound gets sucked up into that catwalk," Hartshorn points out. The original room was designed by George Thomas Howard, with Lake Systems doing the audio contracting work. "But the system has been bastardized since," says Hartshorn. The system currently includes Electro-Voice, EAW, and JBL components in the cluster, power amplifiers by BGW and Crown, and the Yamaha PM2000 console.

The sound booth at Trump Castle's showroom.

This type of show switches constantly from dialog to singing and dancing, and everything in between. The system is low-powered, but adequate to do Broadway shows. At the show I attended, the band was positioned in the house on a riser coming off the side of the stage into the audience. It takes precise mic placement and level control to keep things audible and clear, while at the same time avoiding feedback potential.

Hartshorn accomplishes this by utilizing carefully placed mics on the lip of the stage for dialog, then switching to the singer's wireless mics during musical portions of the program. All of this is augmented with strategically placed shotgun mics under the cluster in niches filled with foam.

Bally's Park Place

This showroom is entirely unlike most of the other showrooms in town: its original sound system (designed by Paul Magill) is still more than adequate in its ability to handle the needs of today's high-powered shows. So many other systems in town may have been fine at the time they were installed, but changing requirements have quickly outdated them.

The system is mainly comprised of Altec 604 loudspeakers with some extra horns, Soundolier C-803's for the under-balcony delay fill, the Yamaha PM3000 console, and Altec amplifiers. Mr. Magill certainly seems to have been able to anticipate the wide scope of the entertainment industry needs here, and Bally's management successfully met those needs by insuring that the original installation was adequate.

Harrah's Marina

Over at the bay side is Harrah's Marina, run by Joe Marchione. He has been there from the beginning, eight years ago. The original system was designed for Holiday Inn, and when Harrah's took over the project right before completion, the entertainment policy changed somewhat to say the least. Unfortunately, all of the contracts had been issued and the work was well under way. Richard Negus had specified a system that was more than adequate for the needs of the Holiday Inn people, but, even though Mr. Negus anticipated as much as possible, he really did not anticipate Harrah's comparatively outlandish stage shows.

Joe has been constantly upgrading the system with the support of local vendors over the years, but, much of the original system remains: speaker systems with JBL, Community, and Altec components; Crown 1200 and UREI 6500 amplifiers; EAW wedges; Klark Teknik equalization; and the Yamaha PM2000 mixing console. According to Joe, the PM2000 has more than paid for itself, but it's time to replace it: "We're taking delivery of a Yamaha PM3000 in January, along with eight TC1128 programmable equalizers."

There were plenty of mic lines run throughout the complex and around the stage to accommodate changing requirements. A Wireworks snake was used, and its modularity allowed for expansion. There are mic panels located at various points all over the stage, which sits right on top of the bay—often wreaking havoc when the salt air combines with the XLR pins. If there is one thing that Joe is glad he got out of the original installation, it's the detailed documentation clearly indicating each and every wire connection within the complex (and there are thousands).

Trump Castle

Like Harrah's, Trump Castle was a last-minute buyout from the Hilton organization. Unlike Harrah's, the Hilton group was attempting to build the biggest and best of all the casinos—especially when it came to entertainment. Steve Gietka, the lead sound tech in the showroom, had the same problem to deal with as many of the other rooms in town—how to take over a new system and put on a huge stage show within a matter of days. George Thomas Howard Associates designed this room, and even though they tried to include every possible feature, envisioning every possible requirement of a creative and artistic soundman is about the most difficult task a consultant can attempt.

There were not enough of the right kinds of mics, direct boxes, and piano pickups. Steve, who had previously worked for Maryland Sound, was used to just reaching over to the large roadcase and pulling out direct boxes, adaptors, etc., and reaching over to the patch bay and plugging in any effect, limiter, compressor, etc., that he could possibly need. When putting on a complex show, a complex number of possible audio combinations are needed to help in the support of the illusion that the producer is striving for. The system features speaker clusters and power amplification by Altec, the Yamaha PM3000 console, Klark Teknik equalization, and two Tascam tape decks: the ATR 60-16 (16-track) and ATR 60-4 (four-track), synched with three Lynx time code modules.

Bally's Grand

The old Golden Nugget, now Bally's Grand, has long had the reputation of bringing in the top entertainment in town. Sammy Davis, Jr., Frank Sinatra, Stevie Wonder, Vic Damone, and

(continued on page 75)

Audio Signal Processing

by Gary Davis and Ralph Jones

In the broad academic sense, signal processing is generally understood as a process whereby the original information undergoes some measurable and controllable degree of change. The original signal may be analog, or may be converted into the digital domain and the continuous stream of numeric datum handled as a series of mathematical manipulations.

Academically, signal processing is usually thought of as a series of mathematical functions first modelled in a large computer (or a micro outfitted with a dedicated DSP board), then later executed in an analog hardware device or digital hardware/software product. Applications include modems; speech communications; radar; satellite; rf; ultrasonic; biomedical; video/image; voice/speech recognition and synthesis; musical instrument synthesis; spectral/function analysis; broadcast and production video; industrial non-contact materials measurement and inspection; geophysical; navigation and guidance systems; teleconferencing; and voice/music enhancement.

When encountering the term "enhancement," most of us imagine a process that may alter the timbre of a singer or instrument. However, in the academic world, enhancement is often referred to in the processing of a signal to increase its signal-to-noise-ratio. While the digital signal processing functions happen in the time domain, their analog counterparts usually happen in the frequency domain (using various forms of filter and amplitude components).

One possible exception to this is the analog "bucket-brigade" and similar delay types that perform frequency-weighted delays, but in the time-domain. Other permutations of signal processing are hybrids of analog and digital circuitry, especially the ever-increasing popularity of the microprocessor-controlled analog circuit. However, as far as as signal enhancement of speech and music signals are concerned, the digital audio method is the de facto standard.

Reverberation

Reverberation is often confused with delay (or echo), especially since some modern signal processors can provide both. Delay refers to one or more distinct sound images (echoes). In fact, true reverberation normally begins with a few relatively closely spaced "echoes," known as early reflections. These are caused by the initial "bounce back" of sound from nearby surfaces. However, as the sound continues to bounce around, the increasing number of reflections blend, creating the more homogeneous sound field we call reverberation. The natural occurrence of reverberation, including early reflections, is shown in figure 1.

The editors would like to thank Gary Davis and Ralph Jones for permission to use material from the Yamaha Sound Reinforcement Handbook.

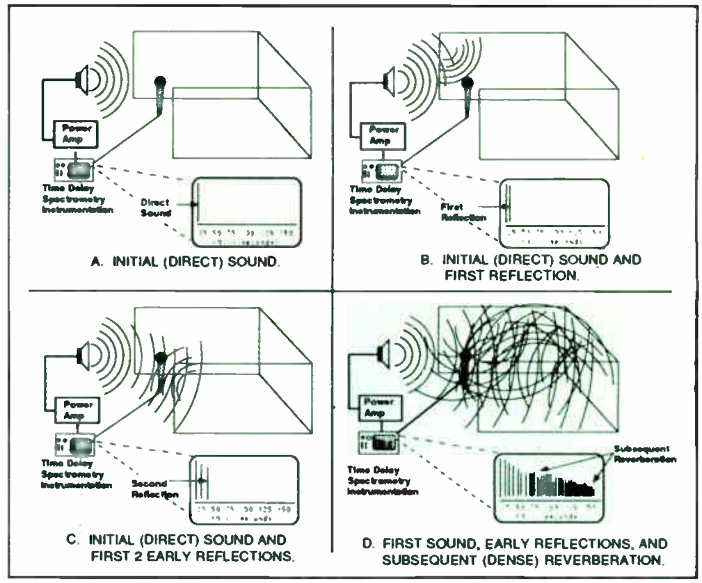


***Figure 1:* natural occurrence of early reflections and subsequent reverberation.**

In theory, if the echoes occurred closer together (in time), they would be equivalent to the early reflections of a reverberant field. We begin to see, then, that there is really a close relationship between echo and reverberation. Those electronic devices which create echo (delay lines) and those which create reverberation share many, many similarities, which is why some such units are capable of producing both effects.

We will discuss some of the methods by which these effects are created, along with a few hints for obtaining the best results. One hint we feel applies to all such effects is this: any effect normally will constitute only a small proportion of the final program; if you don't like what you're hearing, and you've got the mixing console's reverb return level set equal to or higher than the program level, pulling down the effect level may improve the sound considerably.

Digital Reverberation

In the mid-1970s, the first high quality digital reverberators were introduced. They had limited features compared to today's models, and cost upwards of $10,000. While memory costs and CPU (Central Processing Unit) costs have come down recently, it still costs more to obtain a wider bandwidth in a digital reverb. Of course, with the proper design, 20 kHz bandwidth is relatively easy to achieve. On the other hand, it is seldom needed: in the "real world" reverberation usually has significant roll-off at higher frequencies due to the selective attenuation of air at these frequencies. Thus, a reverb with 12 kHz to 15 kHz bandwidth may sound perfectly natural...especially when the effect is blended in with the direct, full-bandwidth program.

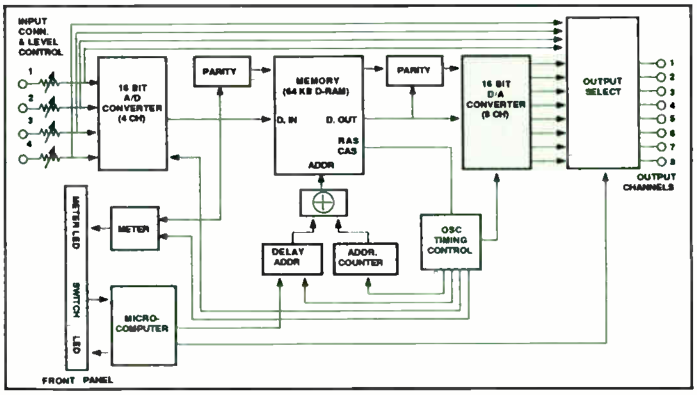
One of the best aspects of digital reverb is that its characteristics can be altered significantly by simply changing the internal algorithm (selecting a new program, in essence). A number of different programs typically are resident in ROM (read-only memory), and some reverbs allow the operator to alter the programs to custom-tailor the reverb characteristics. Of course, all programs operate in different ways, and one of the things that differentiates one model of reverb from the next is just how well one likes a particular program. For example, the number of early reflections, and their relative level and polarity, can have a lot to do with the realism and overall sound quality. Some units provide only one or a few reflections, others may provide from as many as 40 early reflections with precise control over each. Of course, with more possible settings comes a greater responsibility to set the system up properly, which is why preset effects remain popular. Ultimately, there is a degree of "magic" in making a good-sounding reverb.

Some digital reverbs not only enable the user to alter the program, but also enable those altered (or edited) programs to be stored in memory for future use. A number of models also allow for remote control, whereby different programs can be selected, or the effect can be turned on and off, via MIDI (Musical Instrument Digital Interface), SMPTE (Society of Motion Picture and Television Engineers) time code controllers, or a variety of dedicated remote controllers. Additionally, because the technology is much the same, some digital reverbs also offer the user other special effects (phasing, flanging, chorus, echo, gating, and so forth).

Digital Delay

Advances in digital technology in the early 1970s made it feasible to design a practical digital device to delay an audio signal. This device came to be known as a digital delay line (or DDL), probably in honor of the first signal delay. That first delay was conceived about a half-century ago, before the advent of tape recording, and was used for a live radio broadcast. The audio was sent hundreds of miles down telephone lines to another city, then returned via phone lines; the time it took for the signal to propagate along those hundreds of miles of wire constituted the delay time. Digital technology now made this "long line delay" possible without the long line.

In digital reverbs, audio samples are stored in RAM (random access memory) registers. A clock (crystal oscillator) generates strobe or sync signals that cause the memory in each register to shift to the next register in sequence. The signal delay involved is dependent on the number of memory registers available, and the speed of the shift clock. In some units, the selection of longer delay times is available only at reduced bandwidth. This is because a fixed amount of memory is available, and the memory can be used either to store the greater number of samples required to represent higher frequencies, or to delay a smaller number of samples for a longer time.



Block diagram of a 4-input x 8-output digital delay line.

Today, many digital delay systems include a variety of special effects capabilities. By varying the internal clock frequency, the speed at which the signal is "read" out can be changed, which causes a shift in pitch. Often an LFO (low frequency oscillator) circuit is provided to modulate the clock, which produces a vibrato-like regular shift in pitch; usually the LFO frequency (the "speed") and the amount of modulation (the "depth") are variable, and sometimes the LFO waveform is adjustable as well. Some units can be set to "loop" the sound, whereby the input is turned on for a set period of time, and the resulting sound samples are continuously recirculated in the delay memory; the output samples this continuously recirculating sound without destroying it, and the result is an endless repeat.

Other effects include chorusing and flanging, whereby very short delays, which vary somewhat in time, are mixed in with the direct signal to get "comb filter" effects. These effects are discussed in greater detail shortly.

Analog Delay

An analog delay is an effects device which is similar to a "digital delay" in that it is an all-electronic device which temporarily stores audio signal to create a time delay. The analog delay differs in the manner by which the audio signal is stored. Both units sample the input signal, chopping the waveform into thousands of equally timed segments per second. However, whereas the digital delay converts each sample to a number, the analog delay converts each sample to an average voltage value. Instead of using an ADC, numerical (digital) memory registers, and a DAC, the analog delay line uses a "sample and hold" circuit to convert the continuous input signal to a string of voltage values, plus a large number of capacitive storage devices known as BBDs (Bucket Brigade Devices). The reason they are called "bucket brigade" is that the voltage stored in one capacitive memory register (bucket) is "poured" into the next register (bucket) in sequence. Eventually, the sampled voltage reaches the output. The voltage is transferred from one register to another by a strobe signal, much like the digital delay's technology, and many of the same special effects are available in

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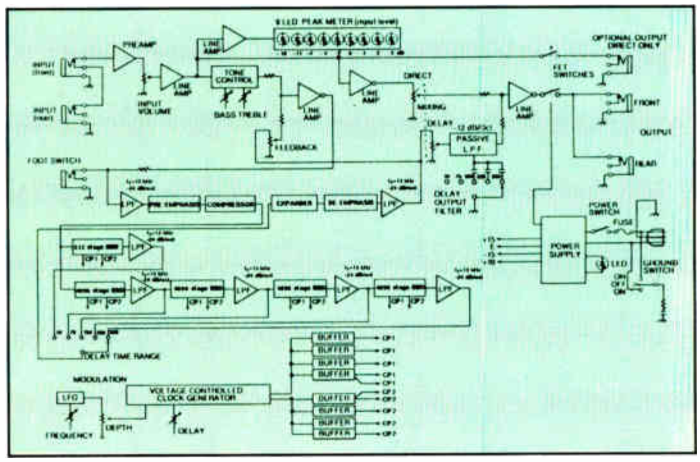
the typical analog delay line.

The typical analog delay line is limited to a narrower bandwidth than the digital delay, and is generally somewhat noisier than the digital delay, although individual models must be compared. Analog delays became popular in the late 1970s because they were less expensive to produce, at that time, than digital delays; they were, and still are, primarily intended for use as musical special effects devices more than for architectural sound delay.

Some musicians claim that the analog delay has a "warmer" or "fatter" sound than the digital delay. Certainly, there are differences in the processing. We suspect that the differences today are as much a function of the specific design of the analog or digital delay as of the basic technologies themselves. In any case, as the cost of digital components has come down, and expertise in their application to audio has increased, digital delays have pushed the analog delay to a small corner of the market.



Block diagram of an analog delay line.

Flanging

Originally, "flanging" was achieved using reel-to-reel tape recorders. Two tape recorders would record and play back the same program, in synchronization. By alternately slowing down one machine slightly, then the other, different phase cancellations would occur. The "slowing down" was achieved by using hand pressure against the flanges of the tape supply reels, hence the term "reel flanging," or simply, "flanging."

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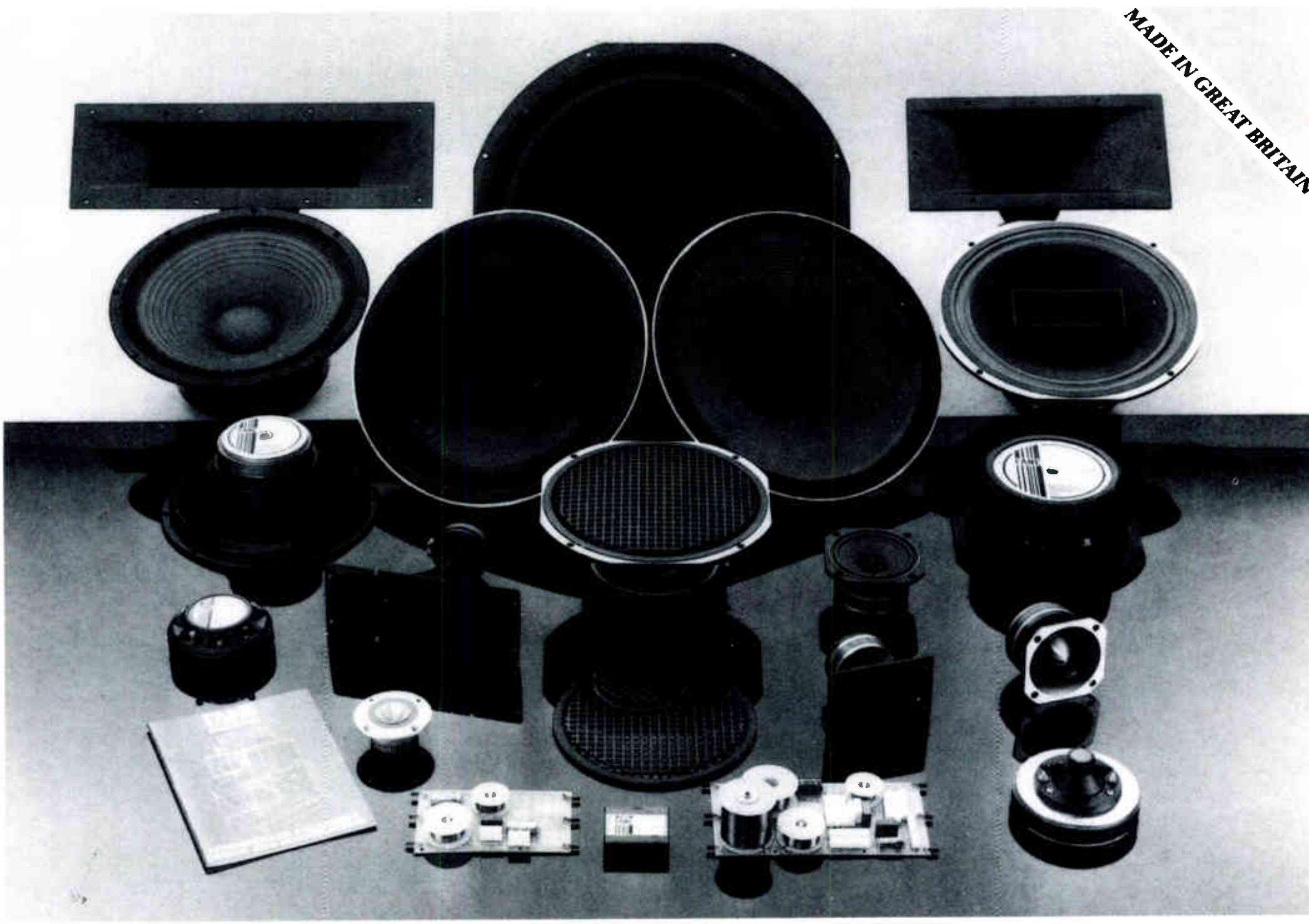

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The result of this alternate slowing down of one machine, then the other, with both outputs electrically mixed, was a series of changing interactions between the two outputs. There would be reinforcement (addition) and then cancellation (subtraction), which give the effect of a sweeping comb filter. The sound can be described as ''swishing'' or ''tunneling.''

Using a pair of hand-tended tape machines is hardly convenient, nor does it produce easily repeatable effects. For this reason, electronic circuitry was devised to automatically create the same kind of effect. If a given signal is delayed, then mixed back with the original signal, the result is cancellation at a frequency whose period is twice the delay time. This cancellation also occurs at odd harmonics of the signal frequency. The depth of cancellation (or reinforcement) depends on the level balance between the direct and delayed sound; an equal balance will produce maximum effect. A sweeping flange is created by continuously varying the time delay, typically with an LFO (low frequency oscillator) to modulate the delay clock. This causes the nulls (notches in response) to sweep across the program bandwidth, alternately boosting and cutting different frequencies in relationship to one another.

If the polarity of one signal (delayed or direct) is inverted with respect to the other, the result is called a ''negative flange.'' If the unit provides for some feedback of output to input, then a more exaggerated effect will occur. If you haven't already guessed, flanging relies upon delay, and that means it is often available in effects-oriented digital (or analog) delay lines, as well as in some digital reverbs.

The highest quality flanging can only be achieved with two channels of delay. This is because complete cancellation is not possible when the delayed and direct signal are always somewhat offset in time. With two delayed signals, there is no ''direct'' signal, but instead both channels are always delayed by a ''basic'' value (whatever that might be), and then the two channels sweep up and down in delay time opposite to each other, with the two delayed outputs being mixed to produce the effect. This affords an opportunity for greater depth of effect. However, it is also a much more costly way to go, so this approach is seldom implemented. (Incidentally, you won't hear a good flange unless the two signals are mixed together electrically; using a stereo speaker system with each channel fed by a different delayed signal will not yield a flange.)

Phasing

Flanging and phasing have a somewhat similar sound, but are achieved in a very different way. A phaser, or phase shifter, is a device which contains one (or more) deep, high Q filters. A signal is split, with some of it going to the filter circuit, and some bypassing the filter. A lot of phase shift is created at frequencies on either side of the filter notch. By sweeping that notch up and down the frequency spectrum, and mixing the resulting signal back with the "direct" signal, a series of ever-changing phase cancellations results.

Changing the relative balance of direct and filtered sound will also change the nature of the effect. In some cases, it is possible to reverse the polarity of one of the two component signals to produce additional, unusual effects. Phasing is popular for guitars, keyboards and vocals.

Because this effect relies upon a swept filter, rather than a swept delay, true phasing is seldom included in a digital delay or reverb system.

Exciters

In 1975, a company called Aphex introduced the first Aural Exciter. This unit changed the signal in such a way that, when part of the Exciter-processed signal was mixed back in with the direct program, the apparent "punch" and intelligibility of the program was enhanced. This was achieved without changing the program frequency balance or gain appreciably. The exciter became popular for increasing apparent loudness of the overall program or of individual parts, especially in record cutting (where the benefit of more apparent level could be enjoyed without reducing the available recording time), in live sound reinforcement (where feedback and headroom were not sacrificed, as they would be with simple graphic or parametric EQ boost), or in broadcast (where greater penetration can be obtained without overmodulation).

Whereas early units were only available for rent, subsequent versions were sold, and later the circuit was implemented on integrated circuit chips and were built into other manufacturers' equipment (today the circuit is being used in special commercial intercom and telephone equipment). Other companies have emulated the function of Aphex's invention, although they use somewhat different approaches to the actual signal processing.

(continued on page 75)

The 85th Convention of the Audio Engineering Society

by Bill Intemann

The Audio Engineering Society will be holding its 85th convention this November 3 through 6, in Los Angeles, CA. Exhibits will be held at the Los Angeles Convention Center, with the Hilton Hotel (just four blocks from the Convention Center) serving as the locale for live sound demonstrations, as well as an extensive program of workshops and technical papers sessions. Dr. Marshall Buck, chairman of the convention, has chosen the convention theme: "A Century Of Technology In The Service Of Artistry."

This year marks the 100th anniversary of two important events in the history of recorded sound. In 1888, Emile Berliner first demonstrated the gramophone (a word he coined) and a recorded disk at the Franklin Institute in Philadelphia, PA. Although it would take several years for Berliner's invention to be commercially available in this country, the technology necessary for playback and mass production had finally arrived. In that same year, Oberlin Smith published his essay titled "Some Possible Forms of Phonograph," the first paper to postulate on magnetic recording. This essay indicated the direction future developments in audio would take, and recent evidence has proven that Smith was in fact the first to have a working model of a magnetic tape recorder.

While the emphasis of this convention will be on the most recent, state-of- the-art developments and applications, the convention theme was chosen to honor past accomplishments as well as current innovations.

Once again AES attendees will be forced to choose from among a plethora of workshops, papers, and technical sessions that will have an impact upon their livelihood and intellectual pursuits. Sessions of special interest to Sound and Communciations readers include:

TV Intercommunications Systems. This session, a workshop, will feature representatives from RTS Systems, Clear-Com, National Teleconsultants, McCurdy and JL Enginnering. Focusing on communications systems for concerts, TV studios and TV Special events (large and small), the session promises to be wide-ranging. with concentrations on identifying the customer's requirements and trying to reduce those requirements to standard hardware. Another planned discussion includes the question of privacy in sub-groups, permitting overiding by higher echelons. Controllers may be PC-based or dedicated, and the panelists discuss the options, along with the choices of building-block modules, performance specification in the real world, and "feedback from the field."

Automatic Microphone Mixers. This workshop on Friday promises to be a complete education. Billed as a "survey of the state of the art," the session features participants from Dan Dugan Sound Design, Altec, Biamp, Innovative Electronic Design, Shure, Industrial Research Products, and JBL. The discussion will focus on the operating principles of different automatic mic mixers and currently available models. Mixers will be compared using standardized tests.

Live Theater Sound. Multiple wireless mics are the focus of the Live Theater Sound Design Workshop. Shelley Herman of Acoustronics Sound is chairman of the Workshop and Gary Rivera of A-1 Audio is co-chairman. Other topics that will be covered include free-floating speaker systems and sound effects.

New Technology. The Digital Studio may not be an everyday concern of the sound contractor, but this workshop, which discusses RAM, hard disk, and tape- based technologies may in fact prove to be a fund of very specific information. Craig Harris of Harris Music promises a panel of users discussing real world experiences of working with digital, including interfacing with the non-digital world.

Milestones. Our editorial not-to-miss choice is this workshop featuring "An afternoon with Jack Mullin." Mullin, the pioneer of tape recording, will be reminiscing and discussing milestones in the history of audio equipment. Chairman of the event is Ron Streicher from Pacific Audio-Visual Enterprises and moderator is Peter Hammar, curator of the Ampex Museum of Magnetic Recording. The AES says this session will be "one of the cornerstones of the convention's retrospective of the centenary of recorded sound."

Technical papers, of course, remain high on the AES agenda, with several new processes being discussed:

Audio Equalizer. A paper on the design and implementation of a phase-linear digital audio equalizer system will be presented by Juan Henriquez, Terry E. Ricmer, and Russell E. Trahan of the Department of Electrical Engineering of the University of New Orleans. Topics covered will include the design of analog Bessel and digital FIR filters for the preservation of phase linearity, and the use of oversampling and decimating techniques. Filter size reduction and flat composite frequency responses are also expected to be described.

Software Package. Robert Moses of Rane Corporation

Intemann is managing editor of Sound & Communications.

is presenting a paper on a software package for audio digital signal processing development. The new software is for audio professionals involved in the design, analysis and implementation of ADSP systems. Pre-AES information says that the package provides a "user-friendly" environment for performing operations such as FIR filter design, IIR filter design, windowing, convolution, FFT, coefficient scaling and quantization. Interestingly, the C language source code is available. And the package will be available at what is called a "nominal" charge. The paper outlines the operation of the package and provides examples.

Linkwitz-Riley. Another paper by a Rane participant is being presented by Dennis A. Bohn of Rane. The paper is titled "An Eighth Order State-Variable Filter for Linkwitz-Riley Active Crossover Designs." It's a subject Rane has been talking up lately, and convention attendees can catch this at 9 PM. (No dinner that night?) The paper will cover performance comparison with fourth- order Linkwitz-Riley designs and transient analysis results for both LR-4 and LR-8.

Underwater. A representative from the BBC is delivering a paper on speech from diving helmets. The concern of the BBC is naturally "broadcast quality" speech, but the paper is expected to discuss the corruption of speech within the diving helmet including spectral distortions. Solutions will be given.

Geometrical Optical Diffraction. Two representatives of Eastern Michigan University will discuss an improved formulation of the optical method, since, they say, experimental evidence has indicated that the optical method is not correct for non-perpendicular incidence.

Radiation From Acoustic Apertures. Earl Geddes reviews optical theories, their implications and examples, and performs numerically a simulation of the acoustics problem and compares the results with the classical optics approach.

Loudspeaker Design—Low Frequency. Kenji Yokoyama of Yamaha describes a concept whereby a compact "low distortion" speaker system has been realized by applying a motional feedback principal to the amplifier. The enclosure also incorporates a newly designed port system.

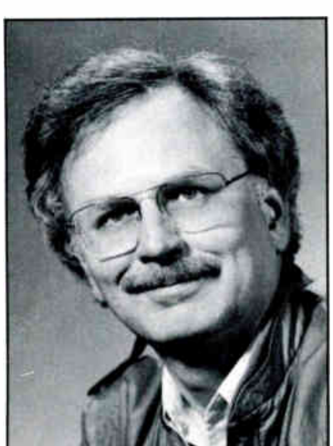
Two special sessions, one Saturday morning, and one Sunday afternoon, should be of special interest to sound contractors:

Speech Intelligibility Session. A special session on Speech Intelligibility will be presented on the morning of November 5. Chairman of the event is David Klepper, and the session promises to be important, with the following speakers:

Don Davis of Synergetic Audio Concepts speaks on "Measuring the Speech Intelligibility of Sound Reionforcement Systems." Davis will have demonstrations.

Thomas R. Horrall of BBN Systems and Technologies, speaks on "Intelligibility vs. Spaciousness of Sound Reinforcement Systems." Horrall will concentrate on multipurpose theaters which must provide good speech intelligibility along with high quality music amplificiation.

Ken Jacob of Bose speaks on "Speech Intelligibility Case Studies from Ten Large Reverberant Rooms." Demonstrations



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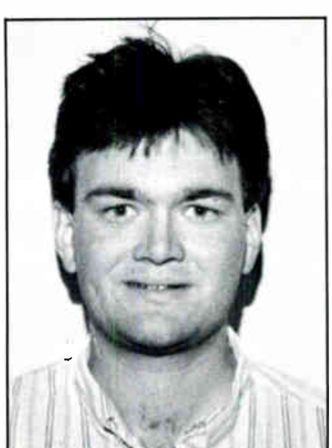
Rane Corporation



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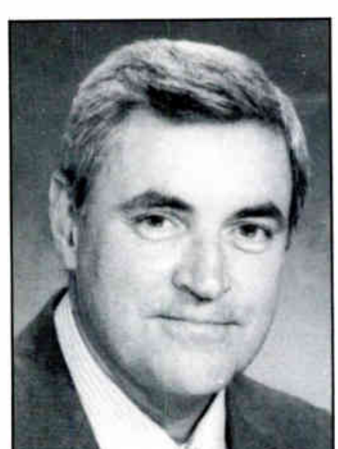

Don Davis
Synergistic Audio Concepts.

Tim Smith
Altec Lansing

of actual measurements and tests will be presented, including reproduction of binaural recordings.

VMA Peutz of Peutz and Associes, Netherlands, speaks on "Speech Information and Speech Intelligibility," covering, among other topics, the distribution in frequency and time of the minimal signals needed in speech information with respect to the sensitivity of the ear, and deriving and understanding earlier, empirically found equations.

Measurements Session. Measurements are always critical, and frequently controversial in the audio world. Gary Sokolich is Chairman of the Sunday afternoon session, with the following topics covered:

Richard C. Cabot of Audio Precision speaks on "Testing Digital Audio Devices in the Digital Domain." Cabot will describe a measurement system which allows testing of digital audio devices in the digital domain, analog domain or a combination of the two domains.

Deane Jensen of Jensen Transformers and Gary Sokolich of Custom Sound Systems will present a paper on "Spectral Contamination Measurement." The new distortion measurement with 110 dB dynamic range shows a spectrum graph of cross modulation products produced by a multi-frequency excitation signal.

"We are using this measurement to study non-linear distortion products produced by frequency dependant group delay (which in itself is a linear error), and to study the non-linear distortion producing effects of overshoot and ringing of an input transformer upon the cross modulation products generated in the audible frequency range from the subsequent amplifier stages," said Jensen.

This new measurement displays linear response errors separately but simultaneously with non-linear distortion products.

Chialin Wu, J. Douma, and G. Bullis of Canetics speak on "Audio Tests via Discrete Frequencies and Linear FM Sweeps." The paper evaluates two approaches in testing the amplitude vs. frequency response of audio electronics equipment. One approach uses discrete frequencies evenly spaced in log space, and the other approach uses linear frequency modulation sweep stimuli in conjunction with FFT analysis.

To sum up: the Audio Engineering Society convention presents yet again a wealth of must-hear sessions, leaving little time for networking and fun. ■

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The Olson Curve (And Other Audio Obscurities)

by Jesse Klapholz

Earlier this year, I had the privilege of being invited to present the opening session at the 6th International Conference of the Audio Engineering Society, held in Nashville, TN. The topic of the conference was sound reinforcement, and the opening paper was entitled, "The History of Sound Reinforcement." A tall order to fill, indeed. The session was divided into three parts: the formal history, some RCA "tidbits," and a short presentation about the RCA electroacoustics lab of the 50s by A.J. May.

The formal portion of the talk covered the invention and evolution of the microphone, the vacuum tube, the amplifier, and the loudspeaker. A majority of the talk consequently highlighted the work of Bell Labs, and touched on RCA to a more limited degree. As a benefit of the John E. Volkmann Library—from which we have been running articles in the past few issues—I discovered some rather interesting stories to relate. These, along with some AES anecdotes extemporaneously presented at the Nashville conference, are presented here for the first time on the printed page.

Who's Who/Watson Letter

Volkmann's obituary stated that he was survived by his wife Dorothy Johnson. Coincidentally, Olson's obituary said that he was survived by his wife Lorene Johnson. A check into the Pennsylvania *Who's Who* for 1939 revealed an interesting fact: Olson and Volkmann were married to sisters.

The question then became who came first, the job or the wife? Volkmann married Dorothy Johnson in 1928, the same year he joined RCA's Photophone Division in New York City, as an acoustical engineer. Olson joined RCA as a physicist. He joined the Photophone Division in 1930, where he worked with Volkmann, and married Lorene Johnson in 1935. Another startling discovery was that Olson held a part-time job as an engineer for Bell Labs while he was going to school at the University of Iowa, from 1924 to 1928.

During the later 30s, Volkmann and Olson lived in the same apartment building (700 Station Avenue, Haddon Heights, NJ). The affair was commented on first by Floyd Rowe Watson, one of the founding fathers of the Acoustical Society of America, in a letter to Volkmann on March 26, 1936: "I was interested to learn that Dr. Olson is your bother-in-law; it makes a congenial family arrangement. Sometime, I hope to meet the whole 'family.' "

Klapholz is technical editor of Sound & Communications.

The Olson Curve

After talking to some of the (retired) engineers who worked with Olson, including Steve Caldwell and Murlan Corrington, the notion of the "Olson Curve" came up. It seemed to most of the guys at RCA that the majority of the speakers they were measuring had some peaks, dips, and resonances that would show up no matter what they did. But not Olson: his curves were always "baby skin" smooth. After some digging, I found what is believed to be the origin of the Olson Curve. In figure 1, the "smoothing" technique applied to the response of the horn can be seen. Also note that the measurements were signed by Olson in 1930, and showed that he was using a Dynes/square centimeter x constant scale for SPL.

Stereo

The following are excerpts from two letters written by J.O. Baker, advanced development section engineer at RCA, to M.A. Rabkin of RCA's Patent Department. They were concerning several Telefunken and EMI Patents that RCA/Europe had options on, and RCA's interest in acquiring and marketing them in the US.

"With reference to the subject docket submitted to me April 19, [1933] it is my belief that the arrangement described by Dr. Hartmann is hardly practical. My reason for making this statement is the fact that if the two banks of loud speakers are separated by any great distance, the person sitting next to any one bank of loud speakers would hear the two sounds as an echo, which would be very annoying....I doubt if the arrangement has any value."

That excerpt is from Baker's letter dated April 19, 1933, so it would appear that he knew immediately after reading what were perhaps the first "stereo" patents that they were of no value—after all, he responded the same day.

However, he did write Rabkin again, on May, 1, 1933: "Referring to my letter of April 19, on the above subject [Telefunken and EMI patents], I was very much surprised while attending the S.M.P.E. Convention to learn that the Bell Telephone Laboratories are working with an arrangement similar to the one described by Dr. Hartmann. In view of the fact that Western Electric are making researches along this line, it may be well to reconsider our decision concerning the subject dockets and possibly file the patent application." Talk about locking the barn door after the horses got out!

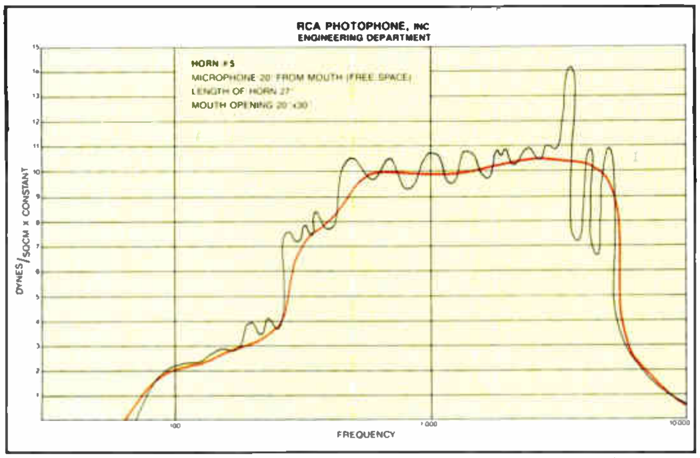


***Figure 1:* "The Olson Curve" (in red).**

Harry Olson
June 5, 1930

Audio 1988

In 1968, at the C.J. LeBel Memorial Symposium, at the 35th AES Convention, Drs. Martin, Pierce, Olson, and Boner were asked to present their respective views on what electronic music, communications, home entertainment, and sound reinforcement might be like in the year 1988. The historic session was appropriately entitled "Audio 1988." Of special interest are Dr. Boner's views on the sound business then, and what he predicted it would be like today.

He said of the typical sound man, "His main occupation may be repairing TV sets, selling drugs, running a parts house, or cleaning septic tanks."

Boner was a pioneer in equalization of sound systems in difficult rooms. He used very narrow-band (5 Hz wide) passive filters, often spending days on the job, tuning and building, tuning and building... It has been said that when asked how he knew when to stop tuning, he replied: "When the customer runs out of money, or I run out of Jack Daniels." He worked many hours in churches and related these feelings about their visions of monetary proportion: "Churches now enthusiastically pay—and brag about paying—$50,000 and much more for their custom pipe organ. They gladly pay $50,000 for custom chandeliers. But they complain about having to pay $175.00 for a loudspeaker in the chandelier, when they could have bought one for $1.25. They also gladly pay major sums for splendid velvet pew cushions for the human posterior, while neglecting the human ears and the brain connected thereto. One can readily establish, in such cases, the ratio between the purchaser's evaluation of the head, ears, and brain versus the posterior. We suggest the establishment of a 'Bottom-to-Top Ratio' analogous to signal-to-noise ratio."

After some far-out Bible quotations (and his interpretation thereof), he went on to make a 21-point forecast of the sound reinforcement system of 1988. [Taken directly from Boner, "Audio 1988." Paper read at the 35th Convention of the Audio Engineering Society, New York, NY, 1968.]

1. The term "sound reinforcement" will have been replaced by a term that is more comprehensive and more descriptive of the art. (The derogatory term "public address system" is already gradually phasing out.) The contemporary custom sound system does far more than "reinforce." It corrects faulty room acoustics. It adjusts bass/treble balance. It supplies early-sound when such sound is missing or insufficient. It furnishes clearly understandable speech in highly reverberant rooms. The art needs a new name that will be fully descriptive of what is accomplished. It will be a respected name, as are the names "telephone," "pipe organ," and "air conditioning." It will not be a name associated with poor sound (or no sound at all), as is the term "p.a. system."

2. Every listening room of any importance will have its own custom sound system, matched to that particular room.

3. The era of the "electroacoustic auditorium" will have arrived.

4. Sound systems will have attained the status of the pipe organ. Like the organ, the sound system will be custom-designed for the room, regulated (i.e., equalized) both by broadband procedures and very-narrow-band procedures (one "note" at a time) to complement the room. In the same way that a manufacturer of a pipe organ will not sell an instrument without voicing, regulating, and tuning it pipe-by-pipe on the customer's premises. Pipe-organ-type finishing and regulating is time-consuming and costly; the 1988 sound system will cost as much as the pipe organ.

5. Electroacoustic genetic-like procedures will be commonplace in 1988. Program material will be created everywhere by such procedures. It will be shaped, colored, and bred at will. Symphony orchestras will have lost their distaste for the "p.a. system" because it will have been replaced by a sound system worthy of the finest music.

6. If capable and dedicated development engineers having youth and vision can be encouraged and rewarded by industry, and not be made subservient to sales and advertising departments, the necessary improvements in sound system merchandise and installation techniques will have been made.

7. Loudspeakers of less distortion than those now available can be designed and manufactured. Presumably, such developments will have occurred. In upgrading and professionalizing sound systems, misleading advertising should be avoided. As the limits of frequency response of equipment items tends to widen in advertisements, but not always in actuality or importance, one solution would be for all to advertise a response from

dc to light.

8. The year 1988 will see electrical inputs from power amplifiers directly into the consumer, supplementary to acoustical inputs. The equalizing of such input systems will be an interesting problem, as will the transducers themselves.

9. The effect of different frequency bands in the sound spectrum on the enjoyment of sound will have been studied and defined. We may expect some startling applications of the results. It is not obvious that existing patterns of energy distribution in instruments, voices, or sound systems provide maximum listening pleasure. Particular attention will be paid to preferred excitation frequencies of portions of the customer's anatomy.

10. Distributed sound systems with sophisticated time-delay characteristics will have fully arrived. We now have luminous ceilings. We will have the "total electroacoustic ceiling," providing both emission of sound and absorption of sound.

11. An all-electrical time-delay system will be in common use. It will be expensive, but so will the custom sound system associated with it.

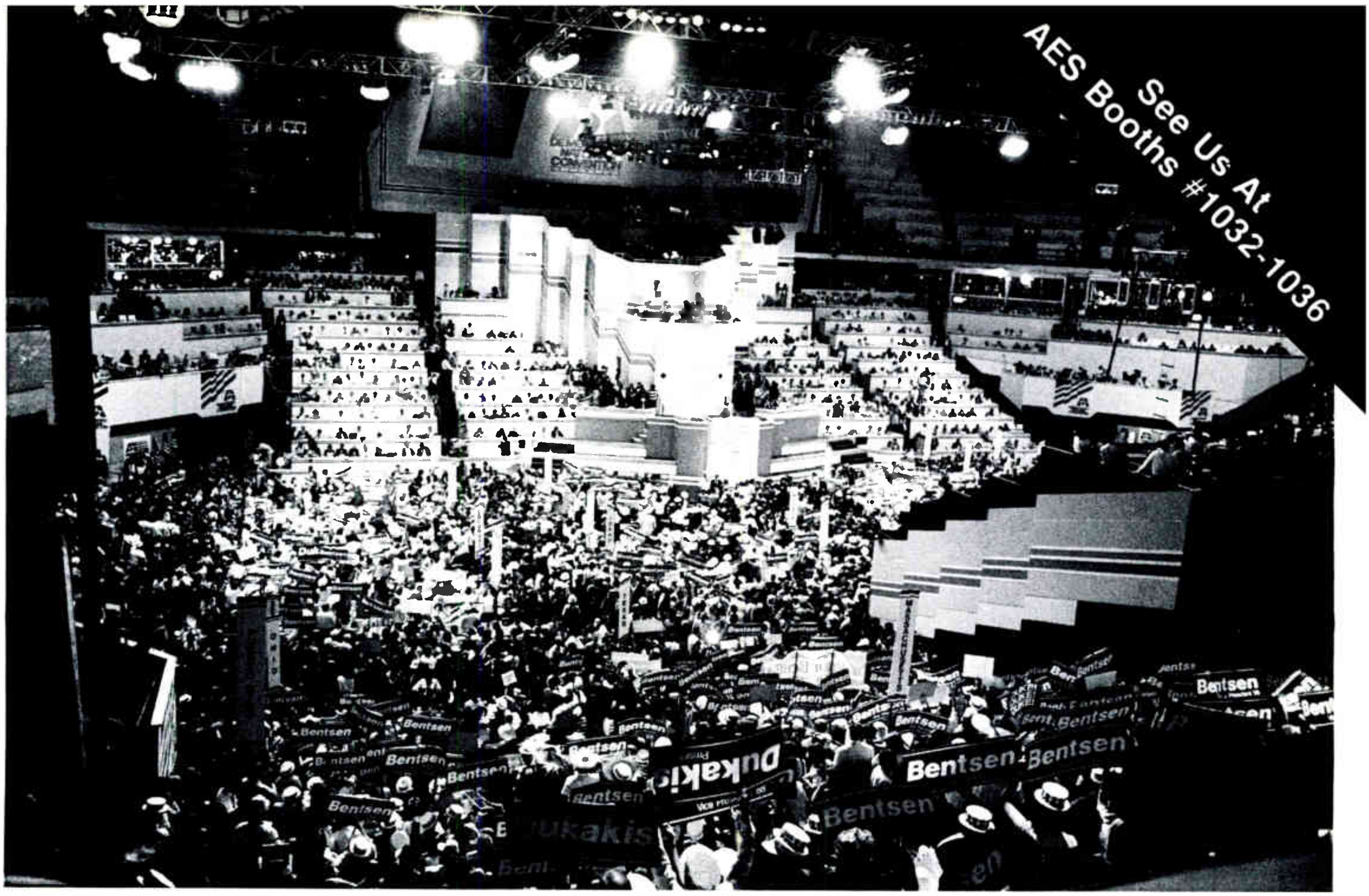
12. The audio engineer of 1988 will have learned to live with solid-state devices of all types. The increase in the cost of custom sound systems will provide a margin within which the manufacturing of more expensive solid-state amplifiers can be made profitable. If this fortunate development occurs, the vacuum tube will almost disappear from sound systems.

13. Microphones of 1988 will have been brought to a quality level to match the requirements of 1988 sound systems. Signal-to-noise ratio problems, variation in frequency response and phase response with time (long-period and short-period) will have been reduced. Directivity patterns will have been improved; leaks in such patterns will have been brought under control. Proximity effects will have been reduced.

14. Fully satisfactory shock mounts for microphones will be available. Otherwise, sound men will have learned not to bolt a microphone to a pulpit, communion table, stage floor, or restroom wall.

15. The inverse-square law will not have been repealed. Acousticians will have fully developed the application of the inverse-square and reverberant- field relations to the maximum benefit of a sound system in a room. Microphones will be positioned in the reverberant sound field more or less routinely. The

(continued on page 76)



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Executive Officer
Democratic National Convention Committee

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Gary Smith
Smith-Hemlon Productions
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Democratic National Convention

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The San Diego Sports Arena

by Mike Klasco

The San Diego Sports Arena is a 15,000-seat indoor sports facility built in the mid-60s. In this genre of architecture, the surfaces were concrete and steel—not exactly acoustically suited for speech intelligibility and music. San Diego Entertainment is the operator/lessee of the arena, and since 1985 has been renovating the facility to make it more competitive with similar venues for both sports events and rock concerts. By June 1986, management was ready to undertake the most complex part of the renovation—the electroacoustics.

The construction management of the renovation was headed by Rudy Palolini, architect and senior vice president of Wagner-Hohns-Inglis, Pasadena, CA. Besides the operator's concerns, Wagner had to assure the lessor, the City of San Diego, that the acoustical/audio deficiencies of the arena would be corrected expediently. As Wagner's acoustical consultant, the author was to devise, computer model, and test the acoustical modifications necessary to provide acceptable acoustics and intelligible sound reinforcement in the arena. Wavelength Systems Inc., El Segundo, CA, installed the acoustic treatment, and new sound system.

The room is 400 feet long by 300 feet wide, with a ceiling height in excess of 95 feet—the internal volume is about 11.5 million cubic feet. Analysis showed flutter echoes, and an average reverberation time of five seconds. The ceiling is an exposed steel deck with parallel 400 foot side-walls, and 300 foot concave end-walls—both concrete. From the echograms (captured by using an IQS 401 FFT spectrum analyzer as a digital storage oscilloscope), the time/energy curves showed strong reflections from the central cluster speakers off the ceiling, and from the upper walls. Refocusing due to the concave end walls was also apparent.

Since the entire upper half of the Arena consisted of acoustically reflective surfaces, using a high Q/constant directivity central cluster to reduce reflections would be a trade-off for reduced uniformity of coverage in the seating areas near the walls.

To correct the room problems, the plan was to absorb and/or diffuse the strong reflections, reduce the reverberation time, and contour its time versus frequency characteristics. Appropriate reverberation times for a rock arena were discussed, as well as how to implement the right amount of absorption and diffraction, and how much of the budget should be spent on absorption versus diffraction. Due to the large internal volume of the arena, reverb times of two to three seconds at mid-frequencies would be a practical goal.

Reduced reverb time would help yield intelligible speech reinforcement by increasing the critical distance. Computer simulations for the reverb time showed that the budget would allow adequate treatment to reach the two-three second mid-band goal, absorb much of the non-lateral reflections and diffuse/absorb the flutter echoes.

The reverberation was measured with a Sigma System RS4000 Computer Audio Workstation. For these measurements we used the sound system to excite the reverberant field, although gun shots, cannons, or impulses (e.g. balloon popping) could have been used as the test signal. For the RT tests the system was adjusted for extended response rather than for maximum intelligibility, or feedback.

The test system generates a pink noise source that is sent to the sound system via computer control. The test signal gates off and the room decay is captured. The energy decay is displayed in 1/3 octave bands. The computer uses a curve fitting algorithm to calculate the actual RT60 by finding a constant slope after the early discrete reflections and before the noise floor "knee" in the decay. To facilitate observations of the spectral decay characteristics, a waterfall-type, 3-D histogram/plot showing energy versus frequency as a function of time is displayed.

A 25-foot concrete wall above the upper seating areas was a primary cause of slap-back echo. To absorb these reflections, a six-foot band of three-inch thick acoustic foam was applied to the 1200-foot perimeter—also contributing about 7200 Sabines of absorption. Ilsonic foam, manufactured by Illbruck of West Germany was used to comply with fire codes. In the U.S., this foam material is sold by Illbruck's subsidiary, Sonex, and the product is marketed as Sonex 1. However, the foam was supplied as a three-inch deep wedge contour by SAMCO of Westland, New Jersey. SAMCO contours the foam with a blade cutting technique, yielding one foot-plus wedge thicknesses. Sonex, on the other hand, uses a hot-wire technique which currently produces a maximum thickness of two inches.

Around the perimeter of the room is an acoustic ceiling suspended 20 feet below the roof, and is built out 25 feet from the outside concrete walls. To decrease reverberant energy here, 10,000 square feet of one-inch semi-rigid sound insulation board was placed on top of the suspended ceiling, and 10,000 of the same two-inch material was installed along the perimeter walls. Because the material is a semi-rigid board, installation

Klasco is president of Menlo Scientific, Berkely CA.

is quick and simple. Since fiberglass itself has no color, the manufacturer can add any color dye to it, therefore, insulation installed in view of the seating was supplied by John Manville in black. This fiberglass added another 20,000 Sabines of absorption.

Rather than use wide-band diffusion, custom units were designed to achieve the combined low-, mid-, and high-frequency absorption. Since the RT60 had to be reduced by three seconds, a further feature of the custom acoustical units was to diffuse any energy that was not absorbed. A wedge-shaped absorber was designed to break-up and absorb non-lateral floor/ceiling reflections, and increase the average absorption coefficients for the ceiling area.

Eight feet long with a pair of two-foot "wings," the wedge frame is covered with three-inch wedge contoured acoustic foam. The inside of the wedge is draped with three-inch thick fiberglass in a "W" shape for maximum surface area. The absorption coefficients are over 1.0 for most of the frequency band due to the multi-plane geometry of this absorber. Even the low frequency absorption characteristic is high for a passive (non-resonant) absorber, due to the eight foot by two foot wide face, and three foot base. By suspending 200 wedges from the ceiling we added 6,400 square feet of melamine resin foam, and 10,000 square feet of fiberglass. Each wedge contributed about 80 Sabines over most of the frequency range, and all 200 wedges resulted in 16,000 Sabines absorption.

Cylindrically shaped Helmholtz resonators two-and-a-half feet in diameter and six feet long were developed to absorb and diffuse wall reflections above the seating areas. By cutting slots into the six foot tube, low-frequency "active" absorption was effectively performed using straightforward Helmholtz resonating techniques. Also, aperiodic dampening of the resonator Q to extend the absorption to an octave was added by placing fiberglass on the inside of the cylinders, and three-inch wedge-shaped acoustic foam on the outside. The resonance tuning of the 100 cylinders are staggered from 100 Hz to 200 Hz. Located about 25 feet out from the walls, the cylinders are suspended from the trusses five feet below the "acoustic" ceiling. The cylinders yield over 4,000 Sabines of wide-band absorption.

Rock concerts place the stage at one end of the arena floor. For this setup a 1000 square foot panel of acoustic foam is suspended from the 90 foot ceiling to reduce the mid and high frequency energy spectrum of the non-lateral "early" reflections. Reducing mid/high-frequency energy content of the early reflections, thereby reduces the amount of higher frequency energy content of later reflections—regardless of the absorption coefficient characteristics of the distant surfaces. The most efficient use of absorbing material is near the sound source, i.e. the scoreboard, and end stage setup.

Before the room treatment, it was impossible to localize the direction of the stage, now the stage was clearly the source of sound. Another subjective difference is the audience applause originates from the audience, while previously the perceived sound source was the roof.

After the installation was completed, reverberation time and echograms were measured. The results exceeded projections, with the mid-band reverb time at two seconds for an empty arena, and the echograms verifying that the flutter echos were eliminated.

The original sound system was essen-

A few of the 100 Helmholtz radiators installed around the arena.

tially unmodified since its original installation in 1967. An old broadcast-style Altec tube console—still working perfectly—was retired, and an eight-channel Ramsa mixer was installed. The output section allows control of the cluster to accommodate the coverage requirements of various events. Ashly electronic crossovers, an Ashly parametric equalizer, and QSC 1400 amplifiers drive the cluster.

The new cluster for the sports arena consists of 12 sets of speakers. Each set uses an EV TL606 vented enclosure for the bass, and a HP series large format constant directivity horn with a DH1A driver for the highs. Originally, the design maintained a tight cluster arrangement. But, the acquisition of a new scoreboard—that was twice the size—required compromising the specification.

The installed cluster comprises four banks of three arrays. In order to cover the entire top-to-bottom seating with a single tier of horns (a physical limitation imposed by the new larger scoreboard), the HP horns were installed for the widest vertical coverage. To reduce the speaker cluster reflections from nearby surfaces, the top of the four-sided scoreboard was covered with three-inch acoustic foam.

The JBL CADP central cluster design program was used, in conjunction with speaker data provided by EV through the Madisound computer bulletin board. EV provides data on their horns

The new central cluster includes 12 sets of Electro-Voice speakers.

but does not provide acoustic data or mechanical design for their bass boxes for the CADP. Data at 2 kHz is the key frequency band for the CADP program—based on the premise that this is the key frequency band for intelligibility.

While this was probably the reason for EV providing data on its horns, the CADP program has (a very basic) cluster mechanical design utility that automatically sketches your cluster if the physical details are on file. As the EV bass boxes are not on file, the complete cluster cannot be automatically drawn showing the full cluster components (i.e. the bass boxes.) Of course, new bass box files can be created, but for the effort involved, better looking results can be obtained using a conventional computer-aided-drafting program (like AutoCAD) or the Bose PowerView program to prepare the cluster prints.

The Bose Modeler and PowerView programs were also used and a comparison between simulation programs is the topic of a future article. The printouts from the JBL program show a partial top view of the arena. Only about 60 of the seating is needed for a simulation when the space and cluster are symmetrical. The CADP program predicted uniform coverage and excellent intelligibility. The benefits of the acoustic renovation of the arena become apparent when the predictions for the ratio are viewed. For example, the "Direct/Reverb for R" simulation shows the ratio of the direct sound level to reflected sound level for the facility when it is empty. Even under these conditions the direct sound is louder than the reverberant field for every seat in the house.

The installation of the speaker cluster actually took longer than the acoustical renovation as the new (unannounced) scoreboard resulted in many unforeseen complications. The arena management had contracted for a scoreboard that was to be the "sister board" of the unit that they were retiring, but the new board turned out to be (to everyone's surprise) the great-grand-aunt! As the new scoreboard was 20 feet by 20 feet and the board it replaced was 10 feet by 10 feet, the original cluster frame originally planned, had to be sent to the "arena museum," located in the parking lot near the trash bins.

Large Union 76 plastic balls (advertising the gas company of that name) had to be moved out the way as they were now blocking the direct sound for 20 of the seats. Rigging used to unfurl curtains to section off parts of the arena had to be raised a few

(continued on page 76)

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CELESTION

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Circle 213 on Reader Response Card

The Ultimate Loudspeaker:

The Wokka "Sweet 1600" Home Construction Project

by Wilhelm Wokka, PhD, DAcDrD.

Why do we have stereo? Have you ever wondered, or have you just accepted this cheap trick? Are you aware of recent experiments in "psychoacoustics" with "stereo" speakers where you can make test tones move overhead, back and forth, etc.? "Psychoacoustic" indeed. This is all "pseudoacoustics" by "psychos," that's what it is. Not only that, how about all this "digital spatial enhancement." Let's be serious here. This is all infantile drivel compared to the Real Thing. . .compared to the Great Works. We have lost so much.

In the Golden Days of Audio, we had only one speaker, and a properly-designed room that did it all. Do you realize that the "mono" recordings from the old days of film and phonograph recording tape masters were able to reproduce a complete three-dimensional, fully spatial ultra-fidelity sound image which was a thrill to experience? These original, full-track "mono" recordings had all the information on them, both space and sound, and when played out of the proper loudspeaker system, they reproduced every nuance of both musical performance and the performance environment. Of course, when reduced to a phonograph record by conniving, commercially-bent opportunists, everything was lost. But the full-track originals: they would put CD, digital, and laser-beams to shame.

In this article, I will reveal the very loudspeaker capable of reproducing it all, as a first step towards the rebuilding of the Great Society of Audio Science. (I long for the day when other Great Men of Science will once again walk the earth, challenging me to perform even *Greater* Works.) You and I, Pilgrim, will now walk the Great Walk towards loudspeaker perfection. We, together, will build the "Sweet 1600 Pleniovonic Variable Expotential Tubeshock Acoustical Radiator." Of course, once built, the stumbling imperfections of all other audio gear will become immediately evident, and a great dread will permeate our industry. However, once the shock wears off, we will rise from the ashes, rebuild the Audio Industry, restoring it to its rightful place. Admiral Perry, roll over.

Theory Of Operation

The Sweet 1600 is elegant and simple, like all Great Works. Magnetic drive levels of 51.7 Tesla provide the proper magnetic force for wide bandwidth. Of course, beyond this, atomic structure collapses and we're left with a black hole (not something you'd want sitting in the parlor, surprising your unwary guests, but properly used, of course, you can have a very effective replacement for your vacuum cleaner). The drive or voice coil has to be exactly 0.007 milligrams per driver, wound from a gold-magnesium alloy which we have to make.

The cones are made from bonded feather slicings (of the common Mexican White Dove) and coated via vapor deposition with a select Uranium/Beryllium/Cobalt alloy, which we have to make. Each magnet will be vacuum-alloyed almco XIV and fortified with a 2500 watt D.C. power supply, which can be purchased. Design equations will be presented shortly. A 40 x 40 array of these components will be "loaded" into a special variable expotential horn with a series of vanes, phasing apertures, and deflectors. Basically, everyone knows that more drivers give better, increasingly more intelligible sound. The 1600 quantity was both derived (to be shown) and arrived at empirically. The following equations will suffice. However, the primary derivations must be kept in confidence, as one would protect the "Dead Sea Scrolls" or "The Book of Love" from misuse and misinterpretation.

A. Magnetic Drive Level

If B = acoustical bandwidth required (MHZ)
D = coil drive diameter (meters)
Mc = coil mass (Mg)
MgR = total moving structure mass (mg)
T = acoustic compression ratio
Vn = variable expotential "n"
$\ominus$ = flux level (Tesla)
$\int$ = center frequency (Hz)
Then:

$$B = 57.1\ \mathrm{ERF}\,(e^{n!})\left[\frac{2\pi\!\int\! Mc}{\Phi D^{1.7}} + \frac{4\pi\!\int\! Msr}{\Phi^{2/3} D^{1.9}}\right]$$

$$+ \frac{T}{(1-j\cosh\Phi)}\left[MsrE^{V_1} + Msr^2McE^{V_2{}^2} + \ldots Msr^nMc^{n-1}E^{Vn^n}\right]$$

Solving for [$\ominus$] and providing T [opt] and other variables (sorry, these are to be kept secret) gives us [$\ominus$] ⁵ 98.8 Tesla. (Not wanting a black hole in the neighborhood, we use the 51.7 maximum allowable.)

B. Number of Drivers

$$N = A\sinh\,(e^{No} + Mc/Msr)$$

$$+ B\cosh\left[e^{-ND} + Mc/2Msr + \left(\frac{4\pi Vn}{c}\right)!\right]$$

Where A and B are empirically-derived constants, that is. A and B were derived as follows: during a joint concert of the Chinese People's Army Orchestra and the Moscow Symphony in the 250,000 seat Polish National Stadium, 9000 special in-

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Dr. Wokka is a Great Man Of Science affiliated with the Philadelphia Medical College of Musical Knowledge, Philadelphia, PA.

strumentation microphones were placed in locations determined by equations 3 and 4.

$$Xn = \sum_{n=1}^{n} \frac{yRn}{n} + P$$

$$Yn = \sum_{n=1}^{\infty} \frac{yRn}{n} + Q$$

Where P and Q are known only to Great Men of Audio Science, 4Rn is the product of the "y" location of Stadium Reflection Rn, and XRn is the product of the "x" location of Reflection Rn. Xn and Yn are the location of microphone "n" in the stadium with (O, O) as the center of the stadium. Of course, these are the standard equations. From equations 3 and 4, and elaborate chart-recorder graphs reduced by millions of graduate student-hours (they work cheap for a mention in a scientific journal) we determined that the optimum number of sources for symphonic reproduction is 1578. We decided to throw conservatism to the wind, opting for 1600 drivers for a bit more punch.

Of course, for "smaller" sounding sources, like these impossible "rock" bands nowadays, fewer drivers are needed, but for human voice reproduction, every one of the 1600 drivers are required. This is not easily understood unless one studies the almost infinite number of "sources" which contribute to human speech, down to the cellular level (my work on an organically "grown" loudspeaker for human speech is to be made available be 1990).

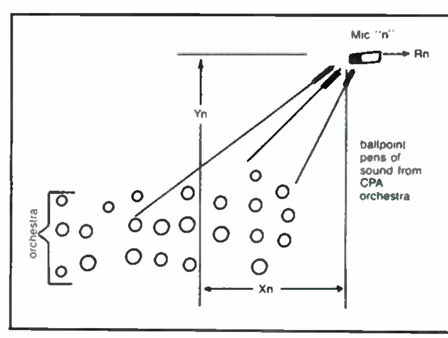


***Figure 1.* Determination of Xn and Yn**

Figure 1 shows how an orchestra and the human voice combine to produce at least 1600 "ball point pens" of sound (a term coined by my imaginative graduate student, Joseph Yovinny of South Philadelphia, PA).

Regarding Sound Pressure Level: for each loudspeaker added, everyone knows you get 3dB more output, netting 4800 dB of gain for the Sweet 1600. Obviously, we need a low-noise amplifier for this. However, rest assured the Sweet 1600 will be loud enough!

This completes the theoretical validation of the design, or as much as I can justifiably disclose to the audio public without danger of misuse.

Construction

The voice-coil bobbin size was determined by a relatively cosmic and arduous semi-empirical, semi-theoretical process. My colleague, Dr. deConsolo determined an optimum size of about 3.5 inches. A sixth sense, (certainly not ego or competitiveness) told me it should be larger, and when the industrious Mr. Yovinny produced exactly the right cylinder on which to wind it (an accident), I was thrilled. I was even more thrilled when the "Contadina" label was removed.

Of course, deConsolo was furious over my bold leap to a larger coil size than his, but True Science waits for no one. The magnetic circuit is straightforward, once the alloy is obtained. Any esoteric foundry will produce the following recipe:

Material/Percent By Mass

Iron, pure/65.007
Byzantium/0.0760
Cobalt/12.076
Carbon/0.00015
Wool, Persian, White/1.076
Tantanium/0.05
Mogami metal/10.66
Lithium/9.77
Potassium/.28485

Uranium, 238 atomic weight Vacuum to 2.0 Tor. 1.000

Heat crucible to 5872 degrees Farenheit. Put in lithium, uranium, and potassium and wait a minute. Put wool in, and watch to make sure a good mix is obtained. Wait 3 days. Throw in the rest. When mogami metal raises to the top and seals the mixture, cool by throwing crucible in nearby lake (check with authorities first). Machine with carbide wheels and grinders (machinists won't care for this,

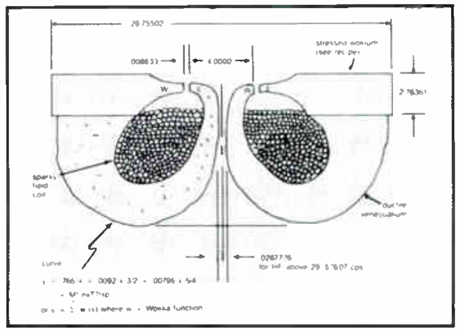


Figure 2. Magnet Assembly

but remind them that they're getting paid). Make pole and plate parts per drawings.

Before magnetization, wind the pole and magnet with 20,157 feet of #75 "Mr. Electro #7" wire alloy, obtained from Mr. Electro Products, Box 75, Beavertown, OR (Ask for "Sparky"). This is great wire, but it's about $25.00 per foot, so be careful not to break it during winding. It can't be soldered or re-welded: you just have to start over. The power supply can be any 12.5 volt, 2500 watt computer-grade supply. Wind the voice coil from the same wire. Oh, I almost forgot: you'll have to insulate it yourself, or get someone to do it. Use good 400 F insulation, and be careful not to break it during this process.

Making The Cone

Put 50 pounds of the dove feathers in melted wax, cool in blocks, and microtone (0.0016-inch plus/minus 0.0001-inch slices) the entire block. Melt the wax out and clean as follows:

1st Rinse in paraffin (kerosene) 200 degrees farenheit.
2nd Rinse in paraffin (kerosene) 200 degrees farenheit.
Rinse in naptha 140 degrees farenheit.
Hot water detergent rinse.
3 hot water clear rinses, 140 degrees farenheit immediately after water rinse.
2 TFE Freon rinse, 100 degrees farenheit.
1 TFE freon vapor degrease.
Dry in oven 3 days, 350 degrees farenheit.

You will end up with a "fluff" of "little spaghettios," or finely sliced hollow cylinders of feathers (you'll have billions and billions). For those true "Audio Jedi" types, sort into groups and mix by diameter per figure 3. Place in distilled water with 1 percent Nutrasweet and strain onto a form per figure 4. Dry, and

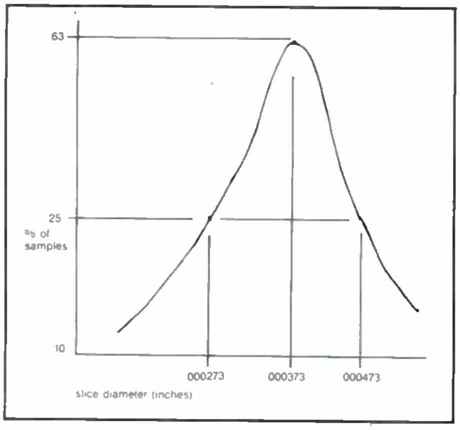


Figure 3. **Diametrical distribution of dove feather slicings**

bake for 4 days. Final thickness should be 0.0015-inch plus/minus 0.00004-inch. Vapor-deposit a 50 percent alloy of Beryllium and Magnesium (obtained from any esoteric foundry you've been dealing with) to a thickness of 0.00026-inch (as seen with a microscope). Don't touch! Don't tap! Assemble with spider, and surround as shown from Hawley Products, St. George, IL. Mix loudspeaker surround damping compound as follows:

Substance/Parts By Weight

Glycerin/6.0
Lanolin/4.5
Cod Liver Oil/2.0
Beeswax/0.18
STP Oil Treatment/2.6
Jello (clear)/4.0
Ambroid Glue/9.0
Chicken of the Sea Chunk Tuna (in oil)/15.0

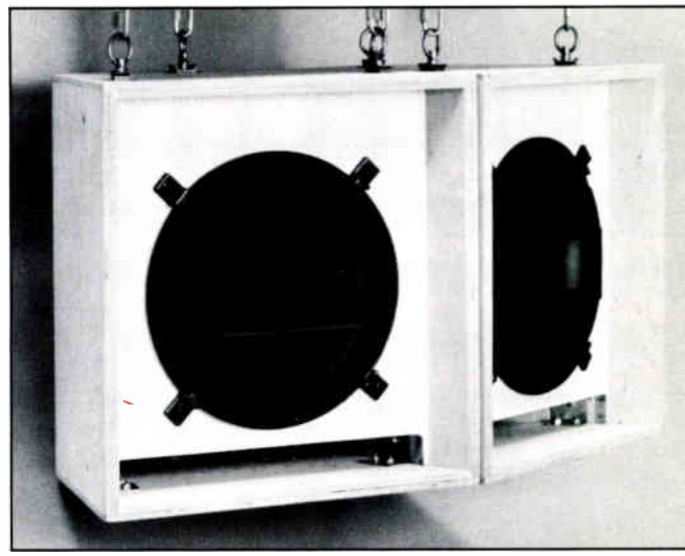
Boil for 9 weeks, or until the tuna disappears and the cod liver oil joins with the STP to smell like an overturned Port-O-Jon after a 4th of July celebration. Apply with pure camel hair brush. Do not get it on anything. If compound comes in contact with the skin, just figure on it being a part of your person from now on. It won't come off (even with the shedding of skin), but it will provide scientifically correct loudspeaker dampening.

Construction of Ultra-HF Fizzer/Pizzazzor

Assemble 5.58 carbon whisker/quartz fizzers and place on three pizzazzors domes (186 on each) per figure 4. Figures 5 and 6 illustrate how deflector and drivers are to be assembled.

(continued on page 76)

Professional Audio Systems PI-122P



Commentary

by Jesse Klapholz

The first co-axial loudspeaker was the now-famous Altec Lansing 604, introduced in the early 40s by James B. Lansing. It was explicitly designed to be a no-holds barred studio monitor. At his own company, JBL, Lansing subsequently introduced both 12-inch and 15-inch coaxial designs. The Altec coaxials (which later included a 12-inch) used a compression driver for the high-frequency, while the JBL's used direct radiator devices. All of these units quickly found themselves in distributed systems of very high quality. In the 70s, two more companies introduced coaxial loudspeakers: Cetec Gauss and Professional Audio Systems (PAS).

At first, PAS introduced a 12-inch coax that used an Emilar driver, exponential horn, Time Offset Corrected (TOC) cross-over, and their own 12" loudspeaker. This unit received quick acceptance in many entertainment applications, mainly because of the amount of high-quality sound the small package was capable of producing. The line was later expanded from the small monitor to include various configurations; eventually a 15-inch coax was introduced. The 15-inch coax soon replaced the Altec 604 in the UREI 813 studio monitors.

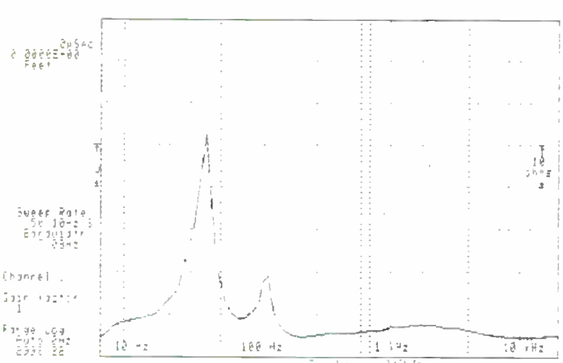


Figure 1 Magnitude of PI-122P impedance, 0-20 kHz, 7 Hz resolution.

A simple observation of the current models reveals a newer horn, of the constant-directivity type. Obviously, as is true of all coaxials, the low-frequencies are reproduced by a direct radiator with little directivity and the high-frequencies are reproduced by a small horn. Therefore, when contemplating the application of these devices, one should consider how they will be heard in the disparate acoustics of various spaces. For example, the intended listener will most certainly be very close to the unit when it is used as an on-stage monitor. However, in a reverberant church, an intended listener may be several times beyond the critical distance in the 500 Hz octave-band (if the room is large, with hard surfaces). This is also the case when using direct radiator low-frequency systems with high-frequency horns, as has become very common with the popularization of Thiele-Small

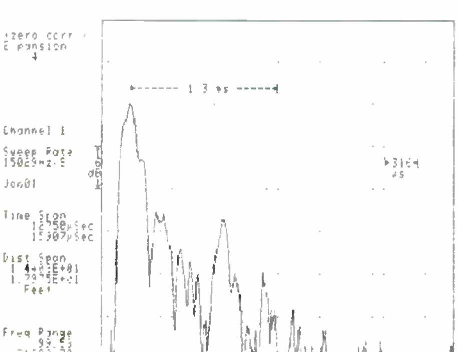


Figure 2 On axis ETC.

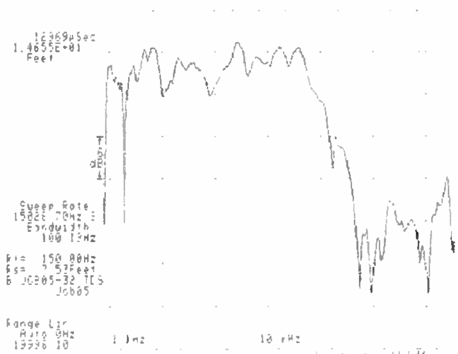


Figure 3 On axis frequency response, 0-20 kHz, 150 Hz resolution.

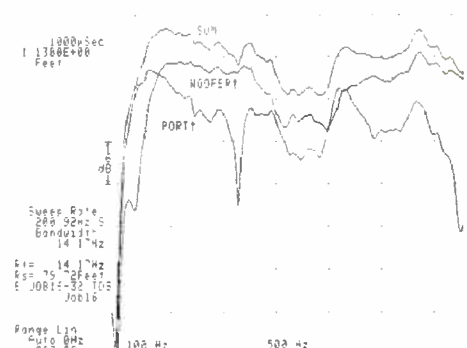


Figure 4 On axis frequency response, near-field, 0-1 kHz, 14 Hz resolution. See text.

low-frequency enclosures.

Previously, one commonly found front-loaded horns using uned bass-reflex ports, such as the Altec Voice of the Theater series and the JBL 4500 series (which featured the notable 4560 "Perkins Box," named after its designer, Cal Perkins). These designs offered some directivity through the crossover region and made an attempt at extending the critical distance in the octave or so below the crossover to the critical distance of the high frequency horn. Nonetheless, high-tech progress has brought us the current wisdom, which is that we only need directivity in the high-frequency horn, and that a Q of two is sufficient in the low-end. In essence, when applying these coaxial devices in reverberant spaces, the same rules apply as when using the popular direct-radiator of larger systems—only the numbers are a little smaller.

The Coax

The PAS Coax design is based on a 12-inch loudspeaker with a three-inch voice coil. When properly band-passed, this device can easily handle 400 watts continuously for extended periods. The speaker has a tapered bore through the magnetic assembly to accommodate a one-inch throat high-frequency compression driver. The taper is the first part of the horn, and as such is exponential. The back-plate of the unit has bolt patterns (for both the two- and three-bolt patterns). Screw-on drivers require an extra adapter. The PAS comes with a relatively shallow Emilar driver: the distance from the diaphragm to the exit hole is about half that of several other popular designs. This decreased distance helps in reducing the amount of delay necessary for the low-end.

The unit is available with either a passive crossover, or an active unit that incorporates a little equalization, preset time-delay for alignment, and limiters for device protection. The focus of this article is on the passive unit that uses all-pass phase delay in the low-end (similar to the crossover in the UREI 800 series of studio monitors). The enclosure is made of multi-ply birch plywood, and uses high-performance rated hanging hardware. The hanging devices double as cabinet reinforcement in that each hardware device is connected to the enclosure wall perpendicular to the one on which the device is mounted. This hardware scheme, along with the multi-ply birch, makes for an enclosure that is both attractive and strong. The unit may be finished with a stain and oil (very easy and inexpensive), or painted. The unit comes with a foam grille, but a hazard-protective metal grille may easily be installed. The enclosure is a trapezoidal shape that encourages array configurations, and minimizes internal high-frequency reflections.

Observations

Our observations revealed indications that the "image" cer-

(continued on page 79)

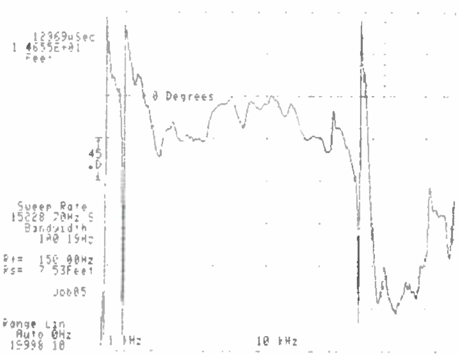


Figure 5 On axis phase response, 0-20 kHz, 150 Hz resolution. Tuned for mid-range.

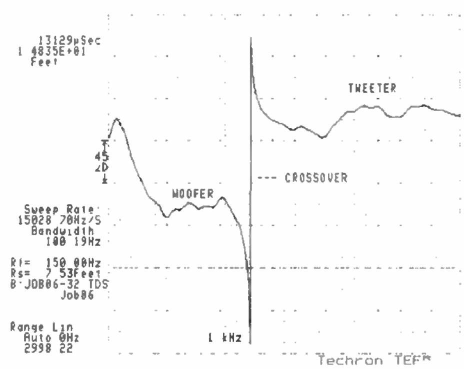


Figure 6 On axis phase response, 0-3 kHz, 150 Hz resolution. Tuned for crossover.

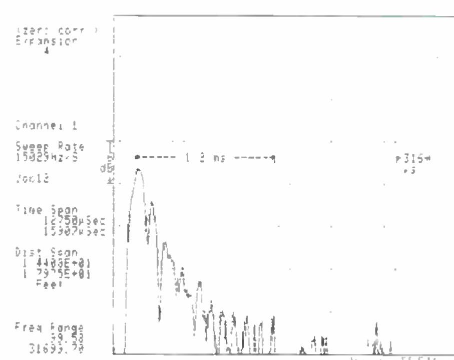


Figure 7 ETC off axis 30 degrees horizontally.

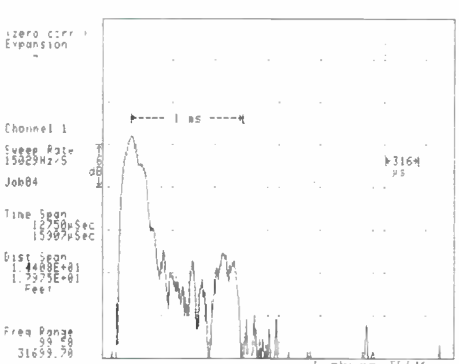


Figure 8 ETC off axis 30 degrees vertically.

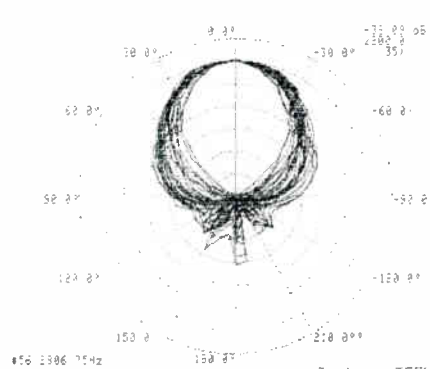


Figure 9 2 kHz composite horizontal polar pattern, 10 degrees/data point, 6 dB/division.

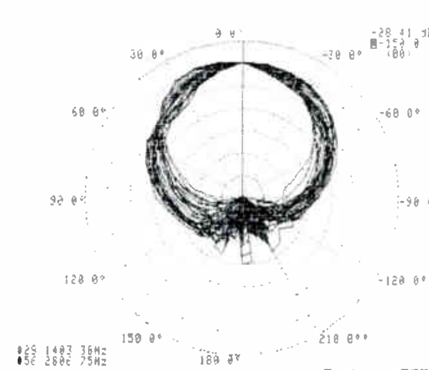


Figure 10 2 kHz composite vertical polar pattern, 10 degrees/data point, 6 dB/division.

People

Industry Movement

Cation named Atlas president; Nordahl moves to Studer Revox

New President of Atlas/Soundolier

American Trading and Production Corporation has appointed Kenneth L. Cation as President of the Atlas/Soundolier Division headquartered in Fenton, MO.

Kenneth Cation President of Atlas/Sandolier

Cation formerly had been President of the electrical-electronic businesses of Prestolite Electric and Sheller-Globe Corporation. His experience also includes 20 years with Westinghouse.

Allen & Heath Appoints New Chief Executive

Allen & Heath has appointed John Ball as Chief Executive. Ball has a strong background in the sound business as well as other related technically based industries. Ball formerly has been managing director of Theatre Projects, a sound and lighting equipment hire and sales company and a partner at K.E.F. Electronics Ltd.

John Ball of Allen & Heath

owner of Studio Supply Company and is a well-known professional in the audio industry for over 20 years. Irby will be responsible for Harrison's SeriesTen, MR-20 and other advanced technology music recording and related audio console products.

Klark-Teknik Appoints National Sales Manager

Klark-Teknik has appointed Sam Spennacchio as national sales manager. Spennacchio will be overseeing the national sales rep network and coordinating all advertising and public relations activities. He will also be involved in product planning and development for Klark-Teknik and its affiliates, DDA, Midas, Milab and Celco Lighting.

Sam Spennacchio of Klark-Teknik

Artec Names Two Associates

Artec Consultants Inc., a New York firm providing arts development, theatre planning and acoustics consulting services, has named two new associates of the firm.

Nicholas Edwards has been a consultant with Artec for the past decade. His responsibilities have included room acoustics design, computer analysis of room acoustics, theatre planning research in auditorium acoustics, computer aided design and modeling of sightlines and room acoustics, noise and vibration control and acoustical isolation design.

Robert Essert, Artec's Coordinator of Acoustics Consulting Services, has been a member of the firm for nine years. His responsibilities have included the coordination of the firm's consulting and research in room acoustics, noise and vibration control, sound isolation and music performance acoustics.

RPL Sales Rep Promotion

Recorded Publications Laboratories, Inc. (RPL) has promoted Donna Azzara Sales Representative for their New York City office.

Donna A. Azzara of RPL

Azzara has been with RPL since 1987. Lee H. Gray, vice president regional sales manager feels, "Donna has shown us that she possesses good management skills and has demonstrated exceptional ability in the area of customer service — one of the most critical facets of our industry. She was the natural choice to represent our busy New York City office."

Tom Irby Joins Harrison Systems

Harrison Systems has appointed Tom Irby as Advanced Technology Product Manager. Irby is the former

Studer Revox America Appoints Vice President/General Manager

Tore B. Nordahl of Studer Revox America

Studer Revox America, Inc. has appointed Tore B. Nordahl Vice President and General Manager in charge of operations. Nordahl comes to SRA from Mitsubishi Pro Audio Group, starting the division in 1983, where he had been President and CEO.

Chief Engineer for Anixter Named

Anixter Bros., Inc. has appointed W.D. Wilkens to the newly created position of chief engineer. Wilkens formerly has worked for Ericsson Cables and has 19 years of engineering and quality assurance experience in the wire and cable industry, including power, mining, plenum, instrumentation, control, electronic, telephone and fiber optic cable.

Aiphone Sales Rep

Aiphone Corp. has appointed Gorden DuVall sales representative for the region covering upstate Illinois and eastern Wisconsin. Duvall will continue to serve as the Airphone sales representative in metropolitan Chicago.

DuVall will be assisted by Marilyn and Paul Bunker of Sunrise Sales, Romeoville, Ill., who have provided the Illinois and eastern Wisconsin area with related products for many years.

Products

More on Linkwitz-Riley; Gooseneck Mics and New Speakers

BSS Introduces Crossover with Linkwitz-Riley

BSS Audio's FDS-310 Sweepable Frequency Dividing System is a crossover that uses 24 dB/octave Linkwitz-Riley filters. Depending on the frequency configuration, the system can be used as a two-way stereo or three-way mono crossover. A switch on the rear panel determines two- or three-watt operation while the front panel indicator designates the crossover's mode. The FDS-310 also offers a "spare" frequency band which is automatically reconfigured as an extra full-range output. This feature allows for the user to drive a separate monitor or flown system directly from the input signal.

Each frequency band on the FDS-310 is equipped with its own set of switches and buttons to control the levels of frequency, polarity, and mute, as well as to indicate signal present

and peak signal warning. Two-color graphics denote the function of each control according to the selected mode. All inputs and outputs are duplicated with both 1/4-inch stereo jacks and 3-pin XLR style connectors.

Anchor Audio Introduces Ensign

Ensign is a table top lectern by Anchor Audio. Ensign's case is rotomolded of polyethylene and filled with foam. A 50 watt MOSFET amplifier powers a dual speaker array. The lectern is equipped with a condenser gooseneck microphone and two phantom powered microphone inputs. Other features include an auxiliary input, treble and bass controls, a line output, and external speaker. Anchor's Ensign also includes a reading light, a 12 VDC

output for use with a wireless microphone, a tray for pencils and pointers, and a storage compartment.

SR Series from Celestion Reinforces Sound

The Celestion SR Series is a compact, full-range sound reinforcement system. The system features an intelligent electronic controller monitor, the SRC1, which provides for protection from thermal or mechanical breakdown.

The SR System has a single-driver design which produces frequency response (50 Hz-20,000 Hz) without intermodulation distortion. A pair of these 8-inch integrated dome/cone drivers are mounted side-by-side in the compact and rigid enclosure. The Celestion SR can handle up to 500 watts RMS or 1,000 watts per cabinet.

When more low frequency response is needed, the modular SR-2 system subwoofer, equipped with a single Celestion G 18Q-400 or B-18/1000 18-inch driver, handles 400 watts or 1000 watts RMS respectively and also can accurately reproduce ultra-low frequencies from 40-150 Hz.

Shure's SM99 Mini Gooseneck Mic

Shure Brothers, Inc. recently introduced the SM99 Miniature Condenser Microphone for gooseneck-mount applications. The SM99 features a one-centimeter precision condenser element and an on-board preamplifier. The mic has a symmetrical supercardioid polar pattern and is designed for sound reinforcement uses that require wide frequency response. Shure's SM99 is equipped with a pop filter for speech and music pickup and can be plugged directly into any surface-mounted three-socket XLR-type connector or be permanently installed by using the included mounting flange.

Burle Introduces Two CCTV Control Systems

Two complete control systems designed for smooth operation of closed circuit television were recently introduced by Burle Industries Security Products Division. These are the TC4500 Series and the TC4500X Series of TransCoax Control Systems.

The TransCoax Control capabilities allow for adjustment of pan, tilt and auto-pan for 24 VAC, 115 VAC and 115 VDC pan/tilts. The 230 V, 50 Hz models have an additional 230 V pan/tilt output. Zoom focus and iris (with variable speed) are also controlled at each camera site. Two on/off switches control latching auxiliaries or auto-pan, and one two-position switch controls two momentary auxiliaries. Additional standard features include 2:1 or random interlace operation (monochrome or color), "backporch" video clamping for hum and noise rejection, and weatherproof NEMA on-site receiver enclosure. The TC4540 and TC4540X Series transmitters, in conjunction with the TC4545 and TC4545X Series receivers, are microprocessor-controlled remote video switching systems specifically designed for use in both control systems.

Audio-Technica's AT4051 Cardioid Mic

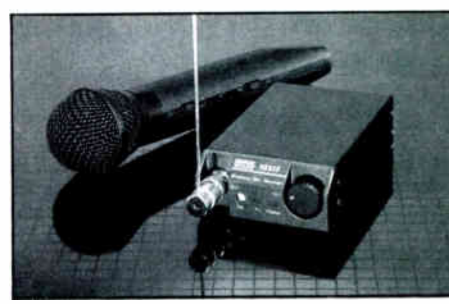
The AT4051 is a cardioid microphone by Audio-Technica U.S., Inc. Engineered for low self-noise and high output, the AT4051 has a frequency range over an extended 20-20,000 Hz, a 144 dB SPL handling capability (1

percent THD), and a -35 dBV open-circuit sensitivity (ref. 1.0 Pa). An optional capsule output attenuator is available to provide up to 159 dB SPL handling capability. Polar patterns are possible by replacing the unit's element with Audio-Technica's AT4049. Standard features also include an integral 80 Hz hi-pass filter, a foam windscreen, and housing unit made of turned brass and plated in black chrome for low reflectivity.

HME Adds Products to Line

HME has introduced a new line of wireless microphones called the 50 Series. The 50 Series mics have two-channel body pacs, newly designed handheld and two-channel switching diversity receiver. The company says that a new RF link improves the capture ratio for drop-out free performance. Other features include HME's NRX-11 noise reduction system, designed for wireless microphones, mic-mute and power switch lock outs on handheld, and RF frequency selection.

HME's System 515 Body-pac and System 525 Handheld are wireless systems for either portable or fixed installations. The System 515 consists of



the RX522 Receiver and the TX550 Body-Pac Transmitter. The System 525 includes the RX522 Receiver and the TX555 Handheld Transmitter.

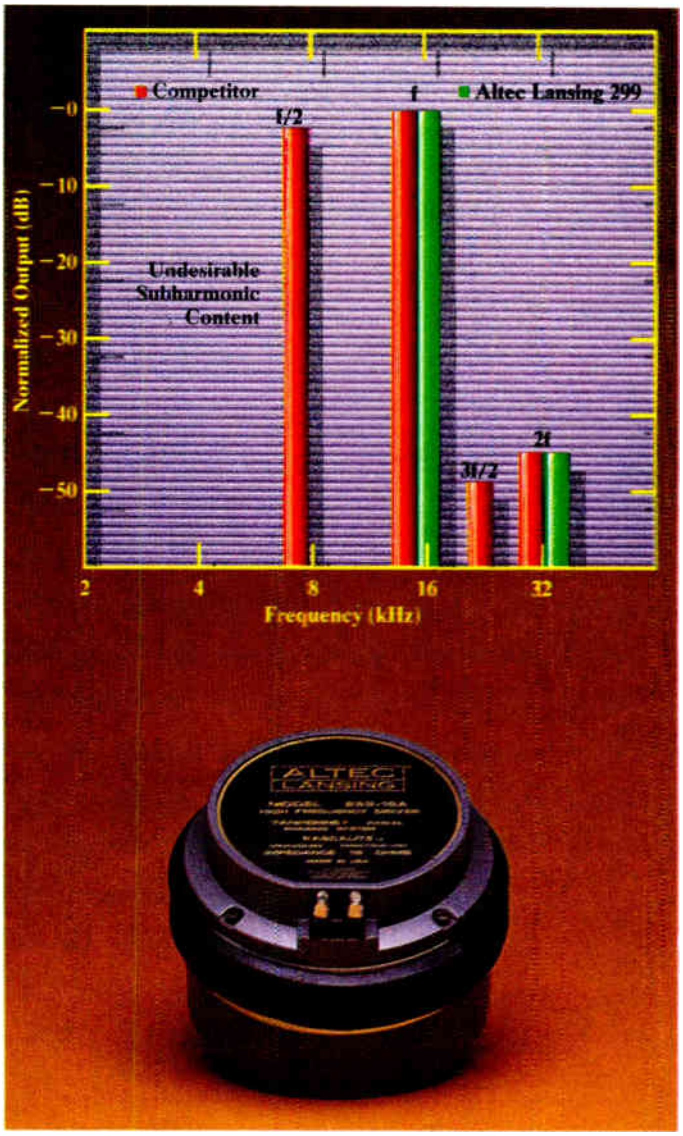
The DN100 Antenna Distribution System allows the user to operate four of HME's RX250 Switching Diversity Recivers in a rack mount and with only two antennas. The DN100 takes up a 19-inch rack and speeds up setup time of multi-compatible systems. The company says a specially designed circuit gurantees against signal loss due to antenna splitting. The DN100 System includes one DN100 Antenna Distribution unit, one AC adaptor and locking clip, and eight RG58 BNC to BNC 4-foot coaxial cables.

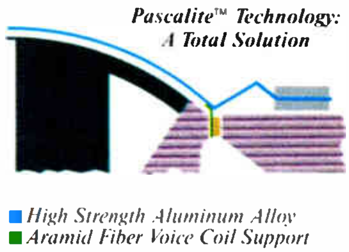
In compression drivers, less distortion equals more accurate sound. Theoretically, transducer sound outputs are an exact copy of their electrical inputs. In real-world compression drivers, distortion must be minimized to satisfy the critical listener. Careful design of the magnetic circuit, voice coil and suspension components helps to reduce non-linearities which cause production of upper harmonics, multiples of the fundamental frequency. Manufacturers' data sheets typically show the level of second and third harmonic components relative to the fundamental.

However, other types of distortions can have greater significance in speech and especially in music. Subharmonic distortion components at one half, one third, etc., of the fundamental frequency are of particular concern in the output of a compression driver. Although the second and third harmonics of the frequency range may be outside the audible spectrum, the second or third *subharmonics* are in the middle of the passband where the ear is most sensitive.

No compression driver diaphragm behaves as an ideal simple piston at all frequencies; some types of "break up" produce subharmonic distortion. At Altec Lansing we solved this problem in the **288 series** of compression drivers. This knowledge was applied in making the careful choices of diaphragm size and material for the Pascalite™ diaphragm assembly in the new model **299** compression driver.

The resulting engineering achievement makes the **299** equal in acoustic output to large format compression drivers using titanium diaphragms, while minimizing the problem of subharmonic distortion.





2.8" Diameter Voice Coil and Diaphragm Assembled with State-Of-The-Art Adhesive in Cutaway One-Half View Mounted in Driver

This graph, based on extensive Altec Lansing research with TEF® analysis, shows the output of a competitor's compression driver with a four-inch diameter titanium diaphragm when driven at 15.5 kHz with a modest input power level of one watt. It also shows the output of the Altec Lansing **299** Pascalite™ diaphragm driver at the same power level. The horizontal axis is frequency, while the vertical axis shows the relative amplitude of the output. At one half the fundamental frequency, the output of the competitor's driver is louder than any of its upper harmonics. This kind of spurious midband tone makes "S" sound like "SHH" and blurs musical definition. *The graph shows the subharmonic content of the Altec Lansing model 299 driver to be virtually unmeasurable.*

The benefit of lower total distortion in a compression driver is more accurate sound, and more satisfied customers and audiences.

Fact.

In Compression Drivers, Less Distortion = More Accurate Sound

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The RP733 Rack-Mountable Power Station is the newest addition to the HME 700 Series line of intercom products. The RP733 is compatable with the entire 700 Series line as well as most other 3-wire intercoms. HME's Power Station has two independent channels with two headset stations that have communication access to one or both channels. The RP733 can power up to 32 belt pacs, with call lights, or up to 100 belt pacs if the call light feature is not needed. Both electret and dynamic headsets can be used.

Also new from the HME line is the EM43 Omnidirectional, Electret Microphone. The mic is designed to operate in RF environments without Audio/RF interaction problems. The EM43 includes a mic clip, windscreen and case.

And HME recently introduced the RM77, a Reverb Mic with reverb circuitry built right into the microphone. The RM77 has an adjustable control that permits the talent to vary the amount of reverb. A three-position switch provides the ability to "mute" the mic, add "echo," or use as a normal mic. A built-in pop filter minimizes external interface commonly heard in condenser mics. The RM77 comes standard with a 3-pin XLR cable, mic clamp, vinyl bag and a gift box.

Sonance Adds to Line

Sonance, manufacturer of in-wall speakers and other architectural audio products, has introduced an upgrade version of its Sonance II two-way speaker system. Maintaining the same

model number and price, the new speaker system employs the same six-inch woofer used in the earlier version, but uses a new tweeter and a newly designed crossover network.

Peavey Announces the UA15 Utility Amplifier

Peavey Architectural Acoustics division has introduced the UA15 Utility Amplifier. The UA15 is suited for a majority of low-power installation requirements. The five-watt capability of the UA15 can handle most audio situations for driving headphones, small monitor speakers, and a broad range of line amplification.

The UA15's level control is a rotary device which is located near the input/output terminal strip. The input may be accessed via phonojack or screw terminals and a pilot light is provided to indicate that electrical power is being supplied to the unit.

DDA Introduces Q-Series Console

DDA has introduced the Q-series console, incorporating many of the features found in its S-Series and D-Series consoles. The Q-Series is an eight-bus console which comes in 16, 24, 32, or 40 input frame configurations. Each input module has eight individual bus assigns, eight auxiliary buses, and a direct output with level control. Each module includes four-band equalization — a pre-set high (12 kHz); a parametric hi-mid (sweep range 470 Hz to 15 kHz); a parametric low-mid (sweep range 70 Hz to 202 kHz); and a pre-set low (70 Hz).

The Q-Series consoles are available in either standard PA or matrix versions, each featuring a large number of available outputs and a variety of signal routing options.

New Offerings From Community

Community has added a sub-compact floor monitor, the wedge-shaped CS28M, to its CS Series II line of loudspeaker systems. The CS28M has the same internal components of the CS25 loudspeaker, but measures only 14-inches high by 15-inches wide by 22.25-inches deep, with a total weight of only 29 pounds.

Based around a two-way bass reflex design, the CS28M is equipped with a 12-inch low frequency driver. At 3,000 Hz, high frequencies crossover to a PZT driver coupled to an exponential horn. Internally protected by fuseless circuitry, power handling is rated at 100W RMS 250W program, while maximum SPL is 117 dB at 1 meter.

Two finishes are available: the standard black carpet covering, or an optional Finnish Oak veneer.

And Community has introduced a new perforated steel grille option that can be ordered on select models of their CS Series II loudspeakers. Available on five models in total, the option is designated by an "SG" suffix added to the regular model number.

On four of the enclosures (models CS70SG, CS52SG, CS38MSG and CS35SG), the perforated steel grilles cover the low frequency speakers only, leaving the molded horn assemblies exposed. On the sub-woofer cabinet equipped with the option (model CS60BSG), the steel grille covers the entire cabinet face.

CS Series II SG models will be provided on a "special order" basis, and will typically require three weeks after an order is received to be shipped.

Milab Introduces D-37 Dynamic Microphone

Milab Microphone Laboratories has introduced the D-37 dynamic cardioid microphone. The D-37 is made of solid brass and features a heavily shock mounted moving coil element for minimum handling noise, built in pop protection and a frequency response of 50 Hz to 20 kHz, with a favorable boost in the vocal/presence range.

Rane's Delivers Dynamic Controller

The DC 24 Dynamic Controller is a

compressor/limiter/expander/noise gate package from the Rane Corporation that features two compressors, two limiters, two expander/noise gates and a built-in 24dB/octave crossover—all in a single-space unit.

The dynamic control functions of the DC 24 operate independently and simultaneously. The built-in crossover feature enables separate band-split processing of high and low frequencies to eliminate "pumping" and "breathing" side effects.

New Combination Speaker From Aiphone

Aiphone Corp. has introduced an eight-watt ceiling speaker/transformer/baffle combination for use with most intercommunications, paging and public address systems.

The SP-5N speaker includes a constant voltage 70-volt transformer with five-wattage tapps, an eight-ohm secondary with press-on connectors and a five-ounce ceramic magnet.

C-T Audio Introduces Disco and PA speakers

C-T Audio Marketing Inc. has introduced the Ohm range of disco and PA speakers in the US. Designed primarily for high-powered applications (such as discos), the Ohms' eight-inch diameter drivers make the systems well-suited for general PA and church applications as well. Hardware is available for mounting the cabinets on walls or ceilings, or they can be placed on conventional speaker stands.

The Ohm MR series provides a choice of 100W or 300W full range speakers, plus a sub-woofer cabinet.

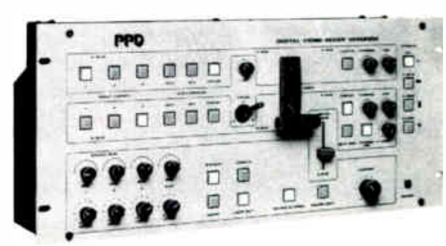
Hitachi Offers New Camera Family

A new family of single chip, solid state color cameras, the KP-C200 series, has been introduced by Hitachi Denshi America. Equipped with a high sensitivity image sensor and a high gain selector of +6 dB +12 dB, each of the series' three models provide pictures with low lag and no burning (even under low light conditions). Model KP-C00, internal sync type, and model KP-C201 with external sync both offer horizontal resolution of 350 lines. The model KP-C205 has RGB output. ■

A Closer Look

Numark Intros Digital Mixer

by Gary D. Davis



Description:

Previously, to wipe or dissolve between two VTR's (Video Tape Recorders), one needed special VTR's, TBC's (Time Base Correctors) for each machine with a common sync gen-

A strong foundation you can build in.

Whether you're a small retail business or a large manufacturing facility—no matter if you're looking for one piece of equipment or a complete system—Panasonic has you covered. With a full line of CCTV equipment to suit a host of different applications.

Like our wide array of B/W and color cameras and monitors—including one of the largest selections of tube and solid-state cameras. There's also a choice of Time-Lapse VCRs and B/W and color Quad Systems and pre-packaged, ready-to-install Mini-Systems. Plus all the Pan/Tilts, housings, switchers and accessories necessary to complement our full line.

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Panasonic
Closed Circuit Video Equipment

erator connected to all the units and the SEG (Special Effects Generator). The PPD VAM2000 combines any three video sources (it accepts three unsynchronized video inputs from VCR's, cameras, or any other combination of video sources), combining them without the need for sync generators or TBC's.

The VAM2000 uses high-speed digital IC's to perform digital freezes, variable rate "strobe freezes," and "freeze-to-the-beat" (of audio) from any one of its four stereo inputs.

A large, long-throw "T-bar" allows complete user control of dissolves and wipe patterns (including vertical, horizontal, and corner wipes). Any one of the three video inputs may be combined with another, or with a background of white, black, or a color chosen (by joystick) from an extensive palette of hues.

Stereo balance and gain controls for each of the four audio inputs are on the front panel, along with an A/B audio buss fader, illuminated mono, stereo, manual fade, and "audio follow video" select switches, located for easy single user operation. In the "audio follow video" mode, the A and B busses for the audio—as well as the video—are faded as the T-bar is moved from buss A to buss B. Separate external video and audio effects loops are front panel switchable for maximum flexibility.

Three video inputs, with three corresponding preview monitor outputs, and three main video outputs, all with BNC connectors, are located on the back panel (along with all the audio input and output jacks). The unit is housed in a 19-inch rackmountable enclosure.

The PPD VAM2000 is designed for video in night clubs, video post-production work, A/V production houses, VeeJay (Video Jockey) systems, cable TV stations, college, industrial, and private TV studios.

For more information contact: Numark Electronics Corp., 503 Newfield Avenue, Raritan Center, Edison, NJ 08837, (201) 225-3222.

Commentary:

I've always felt that those who choose to work in video must sometimes envy their audio-world counterparts. In audio, when we want to mix a couple of signal sources to one output, we need only to make a few quick connections to a single piece of gear, and the job is done. But in video, as the press release points out, even basic tasks like dissolving between two sources have always required several pieces of external equipment.

The precondition that signals be precisely synchronized has traditionally made video systems more complicated than comparable audio systems—and much more expensive. But, my, how times do change! Now, along comes Numark Electronics with an all-in-one video production tool that's no more difficult to hook up and operate than a small audio mixer.

The VAM2000 Digital Video/Audio Mixer accepts up to three video sources and permits you to switch, dissolve, or wipe among them to a single output. It can generate several freeze-frame effects, including a proprietary (patent pending) "freeze-to-the-beat" effect that's triggered by the audio bus. An auxiliary audio input is provided (for an additional "wild" audio track, and both the audio and the video circuits contain effects loops for connection to external processors.

Most importantly, the VAM2000 can operate with unsynchronized video sources. To understand how Numark achieves this feat, let's take a Closer Look at the unit's architecture.

From the front panel, the VAM2000 looks similar to a standard SEG. But this is an SEG with a difference. To achieve clean, stable video effects, the unit incorporates 8-bit Luminance and Chrominance timebase correction. The signal on the A bus is taken as the genlock reference, and the B bus signal is timebase corrected to that reference. This ingenious strategy provides for a full range of basic video production effects, without the need for a complex external "house sync" system.

There are considerable advantages to Numark's design scheme. In practical terms, the VAM2000 allows you to assemble a basic industrial production or post-production system, for example, at a greatly reduced cost. The unit is also a sure winner for club video installations, especially given the "freeze-to-the-beat" function.

As Numark engineer Dewitt Smith was careful to point out to me—much to his credit—the VAM2000 has its limitations. Since the unit takes the A bus as its genlock reference for the B bus, switching between two unsynchronized sources on the A bus will cause a visible glitch on the B bus. As an example, therefore, Smith explained that those wishing to use the VAM2000 as a production switcher should be careful to employ genlockable cameras. This is, at best, a minor limitation, however. The VAM2000 still offers considerable cost savings in such an application, and the problem should never surface in editing applications.

Numark has clearly done their homework. The VAM2000 freeze-frame buffer holds a single 256-line field which is played twice, interlaced, assuring flicker-free stills. The strobe freeze rate is adjustable over a wide range, from 0.5 to 10 freezes per second. Built-in color correction permits the user to match video sources between buses, and individual source outputs allow real-time preview of all three input sources.

The audio section is similarly flexible, and its specifications are im-

Davis is president of Gary Davis & Associates, Topanga, CA, and (along with Ralph Jones) is the author of the Yamaha Sound Reinforcement Handbook.

Crosstalk

A Television Symposium

At the AES Show in Los Angeles

•

Testa Communications will host a series of three talk shows taped "live" before a studio audience during the AES in Los Angeles. Industry figures will meet in symposium to discuss the areas of audio, audio for video and sound reinforcement. These three 1-hour panel discussions will be televised mornings and evenings in convention hotels.

•

Make plans to be part of the studio audience.

Studio taping times and locations will be announced in the next issue or call 516-767-2500 for more information.

pressive for a unit of this type. THD is specified at 0.01⁄ and frequency response at 11 dB from 20 Hz to 20 kHz. The VAM2000 can handle audio input levels from −10 dBV to +4 dBm nominal, with an overload point of 8.0 VRMS. The audio path is stereo, and each input features a separate pan control.

In all, we find the VAM2000 to be a truly exciting new product which has the potential to bring high-quality video effects to a wider market that, until now, might not have been able to afford them. The Numark VAM2000 definitely deserves your Closer Look. ■

Book Review

Room Acoustics in Depth & Upfront

by Steven J. Orfield

Principles And Applications Of Room acoustics
Cremer and Muller, 1982
Acoustical Design Of Concert Halls And Theaters
Vilhelm Lassen Jordan, 1980
Community Noise Rating
Theodore J. Schultz, 1982
Architectural Acoustics
Anita B. Lawrence, 1970

Upon receiving *Principles And Applications Of Room Acoustics*, I discovered that I held in my hands one of the most extensive compilations of information on the subject of room acoustics that I had ever seen, invaluable not only historically, but for use as a current reference, as well as a partial base for design investigations, solutions, and testing. The coverage of the subject divided into two volumes: the first related to geometric and statistical room acoustics, and the second dealing with the mathematics of wave-theoretical acoustics (the more technically complex approach to understanding room acoustics).

The general categories covered in the first volume include geometric, statistical, and psychological room acoustics. Unlike most volumes attempting to deal with this subject area, this is not just a mathematical and theoretical discussion. Rather, it is an exposition of diverse thought within the subject area, interspersed with information on methods of analysis, specific project design, and testing fundamentals related to these theories.

A list of the chapter headings to be found in this volume will give some indication of the range of areas covered: Geometric Room Acoustics, Curved Surfaces, Echo Problems, Sound Reflections With Losses and Phase Shifts, Statistical Room Acoustics, Reverberation, Measurement of Reverberation Time, Absorption for the Control of Reverberation, Psychological Room Acoustics, Possibilities for Judging the Acoustical Qualities of Auditoriums, and The Consequences for the Design of Auditoriums.

Typical acoustical and audio consulting have, for many years, suffered from three problems:

1. Using oversimplified models.
2. Having no understanding of the limits of the model, as to either relevance or correctness.
3. Making the assumption that there is some direct connection between acoustical quantities and human perceptual response.

These two volumes, particularly volume one, investigate many of the basic tenets of room acoustics, from reverberation time to room shape, from echo production and limitation to psychoacoustics. They not only clarify the work of many of the foremost researchers in this field, but they explain in detail the current limits of knowledge in many of these areas.

For those who provide in-depth acoustic testing, or troubleshooting of rooms, or audio systems based on the characteristics of the rooms within which they are installed, these volumes provide some of the most useful information on test process and underlying acoustic assumptions available. For the non-testing practitioner of audio system design, this book is perhaps the best single review of the listener context available. It is without question one of the finest architectural acoustic resources currently available, and Ted Schultz is to be complimented on the high quality of his translation and additions.

Acoustical Design Of Concert Halls And Theaters is a very solid, in-depth work covering the hall design practice of Dr. Vilhelm Jordan, one of the preeminent European acousticians of the last 30 years. This work is extensive in scope and personal in detail, along with a multitude of references to the work of others consulting in this area.

Beginning with the 1940s and 1950s, this work describes the development of subjective and objective criteria for the evaluation of large room acoustical quality (even to the point of discussing electronic enhancement of the performance space). These discussions provide enough detail and explanation to be very valuable to practitioners of room acoustics. This work, like the work of Cremer and Muller, is of extremely high quality, and belongs on the shelf of those interested or involved with the design of large room acoustics.

Community Noise Ratings is the sec-

Orfield is president of Orfield Associates Inc., Minneapolis, MN.

ond edition of one of few good references to the field of environmental acoustics. It deals broadly with the development and use of noise rating scales to characterize human response to noise, for the purpose of evaluating and regulating the control of noise exposure in the environment.

Dr. Schultz, (also the translator of *Principles And Applications Of Room Acoustics*), has provided both a general explanation of the development and application of noise rating scales and has discussed in great depth the suitability and acceptance of these rating scales. In addition to providing the obligatory references to noise studies, Schultz provides a valuable set of correlational information that allows the reader to understand the selection and use of the different rating schemes. He also deals extensively with the specific selection of noise rating metrics based on the noise source type.

The interest within the audio community for this type of information is based on the fact that most good large rooms must have very low background noise levels, and these low levels make environmental noise far more audible. Thus, it is very difficult to deal with environmental noise issues without a clear understanding of the process of evaluating this type of noise. (We seldom complete a major project without at least an informal environmental noise study.)

This volume provides an extensive and in-depth look at the various methods of analyzing event-oriented and density-oriented noise problems, and it discusses many of the psychoacoustic bases for the standards now being used worldwide. It further provides a background of explanation on the use of noise zoning in areas adjacent to noise problems. (Anyone who has been called on to design a large conference room in a building under a flight path will quickly understand the need for this type of information).

This third Elsevier book is again superb in its quality and its scope and, its earlier edition is already considered an essential reference in the field of environmental noise rating and in architectural acoustics, in general. The continuing merging of the fields of audio engineering and architectural acoustics suggests that this is the book of choice for both an introduction and a long term reference to this field.

The final selection reviewed, *Architectural Acoustics*, is a far less comprehensive treatment of the subject area in question, as it is an older volume (published in 1970). It does contain many useful references and basic equations used within the field, and with the current dearth of main reference sources on architectural acoustics, it does provide a basic introduction to the central issues within the field. I would consider this a secondary reference for persons experienced in the field and a primary reference for those new to this area.

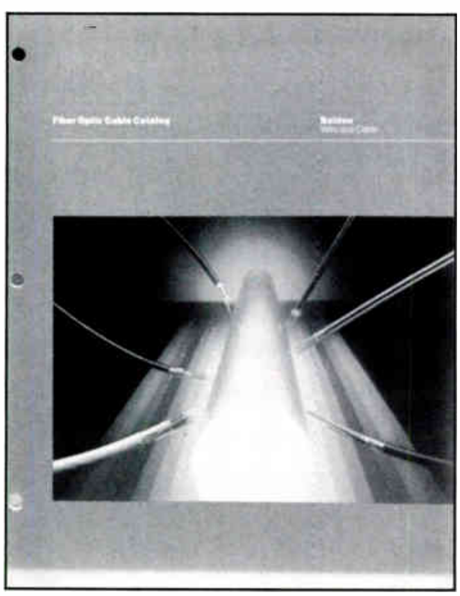
Seldom does one find this many important fundamental references to the same subject area provided by one publisher, and I strongly recommend these volumes to anyone seriously interested in the field of acoustics as it relates to audio engineering. ■

Literature

New Belden Fiber Optics Catalog Available

New Belden Fiber Optic Catalog

Belden Wire and Cable has produced its first comprehensive Fiber Optic Catalog, a 24-page document featuring the entire Belden fiber optic product line. The catalog is fully illustrated and divided into nine sections. Complete technical information including construction details, performance characteristics and standard lengths are provided. A general information section, an index and a complete listing of all existing Belden fiber optic literature are also included.



Microphone and Circuitry Product Catalog From Shure

Shure Brothers Inc., manufacturer of microphones, sound reinforcement systems, and circuity products, is offering a new 25-page catalog on its full line of sound reinforcement, recording, and broadcast products.

Leader's New Catalog

Leader Instruments Corporation has announced the availability of their new full color test and measuring instrument catalog. The 96-page color catalog includes a glossary of oscilloscope terms and application notes on such topics as digital and analog oscilloscopes, video products, programmable RGB video generators and frequency counters. The catalog also has photos and specifications on more than 100 products including 15 new instruments.

Mountain Bell on Top of the Rockies

Mountain Bell, in an article in Outside Plant magazine, described a job of replacing and installing a surface-laid optical cable down the face of Cheyenne Mountain, near Colorado Springs, Colo. Mountain Bell also replaced a copper-cable connection to an "antenna farm" at the top of the mountain. The article covers all the installations and protection devices used for this job.

L.A. Traffic Control Described in Reprint

The Los Angeles Department of Transportation (LADOT) is using Corguide optical fiber in a trunk that links a city hall computer-control center with traffic-signal controllers. The system is described in a reprint from American City and County magazine. In the article, "Controlling Traffic with Fiber Optics," LADOT decided to use an optical-fiber system primarily because of its reliability. Fiber's other advantages over copper-based media are also discussed.

General Machine Products Co., Inc., New Catalog

A new color catalog providing details of over 140 tools and accessories for use in aerial utility cable installation is now available from General Machine Products Co., Inc. The GMP Aerial Catalog includes descriptions and pictures of such standard utility implements as automatic cable lashers, lashing wire, cable blocks, pulleys guides aerial handlines, work platforms and related tools.

Fusion Splicing Techniques by Corning

In an article in Outside Plant magazine, Stuart Saikkonen, senior product engineer for Corning's Telecommunications Products Division, reviews the theory behind fusion splicing and procedures for using manual and automated equipment. He also discusses how to solve splicing problems and presents new information and test results from the lab.

New Catalog From the Master

Master Bond Inc., has now made available a two page application selector guide on thermally conductive, electrically insulative epoxy systems. Viscosity, gel times, service operating temperature ranges, thermal conductivity and application recommendations are listed for 15 Master Bond grades. Both two component and single component systems are described.

Leasametric's new Test-Equipment Product Guide

A new, expanded Test-Equipment Product Guide has just been released by Leasametric. The free guide contains more than 175 pages of illustrated and easy-to-reference information on the broad line of test equipment the company offers for rental. The guide comes complete with product indexes and a manufacturer's index — along with product specifications and selected information. ■

CONSULTANT'S COMMENTS

(continued from page 13)

too loud. If people start moving further and further away from the speakers, it is too loud.

3. If the crowd starts thinning prematurely, consider that the sound level may be a major contributing factor. If people are leaning over and shouting into one another's ears, the sound is too loud.

4. If you can't hear the people coming up to tell you to turn it down because the periodic howling has temporarily deafened you, perhaps it is time to back off.

5. If you notice blood running out of the ears of the other band members, that's the clincher. Grab that master fader and lower the setting pronto!

Seriously, we all have a responsibility to make sure the powerful tools we have provided are used with care. We do not want to suggest installing smaller amps and/or fewer speakers: the answer lies in education. Those of us who specify, sell, install, and/or train people in the use of sound systems have an obligation to educate the sound system user about what levels are appropriate, what levels are dangerous, how threshold shift and distortion affect the perception of level, and how to know when things are too loud regardless of the actual SPL. We should explain that loud stage monitors may mask even louder house speakers, and that the audience may not be enjoying the same kind of experience that exists on stage. We should impress upon the soundmen and bands out there that the purpose of the sound system is not self- satisfaction, but audience satisfaction. Respect for the audience is the mark of a truly professional musician or soundman. Anyone in this trade who doesn't care about his/her audiences's experience of the event will ultimately fail.

An additional concern for contractors is the possibility of intervention by OSHA (Occupational Safety and Health Administration). Regardless of the current administrations's apparent reluctance to enforce the exisisting code, this political climate won't last forever. Sound industry professionals need to be made aware of the potential for federal and local government clampdowns if they continue to indiscriminately pursue the "balls to the walls" approach.

There have been plenty of skirmishes with the audience, and a number of major battles with local authorities carrying SPL meters. Nobody has yet won or lost. Whether or not the Big Guns (feds) call out the heavy artillery (absolute SPL regulations), our audiences and bands deserve better. Let's call a truce in the SPL War right now. ■

THEORY & APPLICATIONS

(continued from page 16)

In terms of current system use, there are a number of actions in the design of a teleconferencing room which may tend to reduce the interaction between the room and the audio-visual system in the production of echoes:

1. Microphones which are narrow in acceptance angle will tend to pick up less of the received signal for retransmission.

2. Loudspeakers which do not have the microphone system within their coverage area will provide a more limited signal for reproduction.

3. A ray diagram of the conferencing facility will quickly point out the most sensitive areas of reflection where detrimental paths between room surfaces; and the loudspeaker-to-microphone path are, thereby indicating "room effects" areas to be treated.

4. Equalization tests on the microphone system, speaker system and room will allow the audio designer an ability to desensitize both the room and the system to this effect.

5. Room reverberation within the audible range of the signal should be examined and limited, especially within the central voice frequencies, as noted by Articulation Index standards. (ANSI).

While small rooms do not generally need a careful look at audio performance and absorption spectra, teleconferencing rooms often exhibit anomalies which are more easily remedied by the techniques used in large room audio (Q measurement, coverage calculation, reverberation calculation and intelligibility analysis and feedback analysis).

It would be very helpful, in the future, if the manufacturers of this type of equipment would supply advice and documentation as to the optimum uses of their systems in terms of these geometric room acoustics and audio system design variables, along with explanations of the significant benefit of each action which is recommended. It would also be useful to look at the energy-time-curves of echoes in these rooms, along with the local frequency response of good and bad systems at the listener position. Finally, it would be helpful to look at plots of the localized sound field at different points within these rooms, under normal and problem conditions, in order to better understand the room-system interaction. ■

CASINOS

(continued from page 28)

others all place unbelievable demands on the system. These superstars expect excellence in the venues in which they perform, and, evidently, excellence is what they get at the Nugget turned Grand.

The sound system features a Yamaha PM3000 console—the first unit sold in the US! Its serial number is 008 (the first units—001 through 007—were built as prototypes). Recently they added a Ramsa 40-channel monitor board. The original speaker cluster proved inadequate. Lead Technician Mark Richman and his crew added EAW 3-way side clusters, and rear-mounted speakers with delay for the rear seating booths and VIP sections, providing flexible coverage. (The rear speakers are dialed up for comedians, for example, where the spoken word is critical. Then it is lowered for the musical headline act—here the stage and front clusters amply fill the entire house.) The acoustics are dead, which means there are no problems with reverb or slap: the engineer has complete control of these parameters through outboard processing, and the room does not interfere with the mixer's intentions.

Mark realizes the importance of preventive maintenance. "Our operating equipment is 100 feet from the ocean. Nearly 100 percent of our critical failures are due to oxidized input/output jacks and dirty potentiometers from the harsh environment—the salt air." Therefore, equipment gets an overhaul and cleaning every six months. All of the electronic equipment is left on, except for the console because of heat build-up. The sound booth is covered with a sealed tarp between shows, and filtered air is pumped into the area. The fan filters, filthy after a week, are cleaned often. Mark has a shop full of repair equipment and spare parts—scopes, meters, electronic components, and so forth—enough spare parts and electronic components for every piece of equipment in the house. If something breaks down, a spare is immediately put in its place, the faulty item is repaired (always in-house—and usually within two hours), and nothing is down for more than a day.

Circus Maximus

Perhaps the most attractive room in town is the Circus Maximus at Caesars. There is an atmosphere of total professionalism surrounding their presentations that is immediately apparent upon entering the theater. The arrangement of the side clusters here makes the most sense of anything I have seen (and heard) in this city. There is no comparison to the awareness and intimacy achieved by having the audio come at you from eye level, as opposed to 65 feet over the stage. Paul Swensen and his crew recently installed a new system that is as good as anything this writer has ever experienced. ■

SIGNAL PROCESSING

(continued from page 37)

The Aphex Exciter works by splitting the input signal; part of it goes direct to the output, and the other part goes through a high pass filter network and a harmonic generator, after which that processed signal is mixed back in with the "direct" signal at the output. This processing generates frequency-dependent and amplitude-dependent harmonics. Newer circuitry has increased the sensitivity to amplitude changes such that the system will steepen a wavefront for a sharper apparent attack. This is done primarily with even harmonics so that the positive-going peak is accentuated. Interestingly, the psychoacoustic effect is much greater than any single specification, measured with conventional instrumentation, would indicate.

The key to proper use of any exciter is to use it in moderation. Only a small percentage of "processed" signal should be mixed back into the direct program feed.

A number of other companies have introduced processors which are supposed to perform similar signal manipulation, although Aphex claims to own the term "Aural Exciter," and reserves the rights to their specific process.

What To Look For

With any special effects processors, but particularly with flangers and phasers, it is important to obtain the right unit for the job. Devices meant for direct connection to a guitar may be too noisy, and may operate at the wrong levels for insertion in the signal processing "send/receive" loops of a mixing console. Some of the differences between units relate to the available control of the effect... how deep it can be, how "wide" it can sweep, whether there are manual and automatic settings, remote control capability, etc. These differences can be explored by looking at the front and rear panels.

Some of the more important differences relate to the actual performance, which can only be determined with listening tests using program material which is the same or very close to that material with which the device will be used. Sometimes it seems that an effect which sounds good with a guitar may sound awful with a piano, or what sounds good with a voice may not be audible with a guitar, and so forth. You may be able to make a correction with the available controls, but sometimes a given unit may simply work better than another for a particular application.

This article's scope has concentrated on the popular embodiments, while almost any piece of electronics can be used creatively for effect. To wit, noise gates, preamps, level controllers, filters, etc., have all been used in many effects applications. In the early 70s when very little was available in portable effects, touring sound companies had to rely on their own ingenuity to one-up the competition. One of the tricks they came up with was to set the mic input right at the edge of clipping, this "controlled" distortion, thus, added "punch" to the vocal. The techniques of flanging, and other various ad-hoc methods of modifying musical timbre have been classically developed using concepts understood through musical ability, and/or knowledge of musical acoustics, and psychoacoustics.

These concepts, and the products which they ultimately produce, are important elements of sound design finding their way into the specs we bid on, the performances we must provide audio illusions for, and the systems we design for today's venues. ■

OLSON CURVE

(continued from page 44)

listener may in some cases be placed within the inverse-square field of his loudspeaker, although there are difficulties.

16. Fully automated feedback-control procedures are now possible, both in broadband and 5-Hz band embodiments. Such systems will be in use in 1988. However, as for the pipe organ, there will not be an automatic substitute for the experienced person, with a good ear, who does the work in the customer's room. All rooms will continue to have geometry and, therefore, an array of resonant modes controlling the acoustical result.

17. The market will have various "black boxes" which automatically extend the capabilities of custom-equalizing by important amounts. The improvement afforded by these items will depend on the precision of the equalizing previously accomplished in the room by a capable engineer.

18. New sources of training of acousticians and audio engineers will have emerged because of market pressure. If universities continue to direct their curriculums away from hard-nosed room acoustics and audio engineering into more esoteric fields, training substitutes for such higher education will be found. Apprentice training, on-the-job training, special long-term and short-term institutes are possibilities. Finding competent teachers with actual experience on the job who are well-versed in the considerable literature and lore of the audio art will not be easy.

19. Audio engineers will do a far better job in leading the public toward custom sound systems than they are now doing. The musical instrument, stereo, and high-fidelity industries have done a better job than has the sound system industry. Frequently, one can see a sound dealer selling a customer a splendid, expensive stereo outfit for his home and, concurrently, cutting both the quality and the price on a sound system for the customer's church.

20. The pyroacoustic loudspeaker will be in a production. If the supersonic jet is permitted to disturb the peace, then the courts will presumably find for the pyroacoustic loudspeaker. The future of this unusual loudspeaker in evangelistic churches which preach hellfire and brimstone is obvious.

While it is amusing to sit back and take a light-hearted look at ourselves and the business, these few departures into audio obscurity in no way intend to poke fun at some of the great individuals who started modern-day electroacoustics. With all the seriousness that goes on in our business, it is refreshing to know that even the greatest and most serious of minds had their fair share of fun. Enjoy! ■

S.D. ARENA

(continued from page 49)

feet as the cluster height had to be moved upward to maintain clearance from the playing field due to the increased scoreboard height.

While the CADP program does not show shadowing, the Modeler does, and this was helpful in predicting if the new location would keep us out of trouble. A small time-delayed secondary cluster for under the scoreboard to cover the playing field had to be redesigned as a distributed system as the bottom of the new scoreboard rotated and could not accommodate speakers.

Eventually the cluster was installed, aligned, equalized, and otherwise tweaked. The goal of a house system capable of intelligible and aesthetically pleasing sound was finally reached. Since these electroacoustic renovations, every concert and sports event has performed to the liking of previously dissatisfied sponsors and spectators. The project has unanimously been acclaimed successful by management, designers, rock groups, and attendees. ■

DR. WOKKA

(continued from page 57)

Final Calibration

Vacuum (2 Tor) Input impedance must match the curve of figure 7 within the tolerance window. Pull off findings on the field coil to adjust as shown. If necessary, add or remove extra fizzers and dampering as shown.

Frequency response (calculated from three measurements made in a Wokka Standard listening room) is obtained as follows: place microphones at a, b, and c and measure the "grazing incidence" frequency response with a very fast chart or horizontal speed and very slow vertical speed. Make 300 measurements in each position.

(continued on page 79)

Klark-Teknik Installations

Philadelphia-based contractors Pierce-Phelps, Inc. have installed a Klark-Teknik sound system at the new Gwinnett Country Justice & Administration Center in Lawrenceville, GA. The complex contains 14 courtrooms with sound reinforcement, an auditorium with a complete audio/visual system and a full TV production studio.

New York City's Radio City Music Hall was the site of another KT installation in a sale arranged by Allied Broadcast of Richmond, IN. The Music Hall purchased three KT DN300 1/3-octave mono EQs for installation in the house cluster system. Eddie Santini, the sound engineer for Radio City, was in charge of the installation.

In Mexico City, singer Lupita D'Alessio is using two DN300 EQs and six DN360 dual 1/3 octave EQs in the house and monitor sections of her touring system. Rob McKinley, co-owner and general manager of L.D. Systems in Houston, Texas, handled the sale.

QSC Goes to America's Cup

QSC amplifiers supplied the power for the live sound updates at the recent America's Cup challenge in San Diego, CA. San Diego's Seaport Village was transformed into The Pepsi America's Cup Village where race spectators gathered to watch the race via a Diamond Vision video screen moored in San Diego Harbor. Sound Image , a sound company based in San Marcos, CA, set up the temporary sound system.

Five speaker towers were placed along the Village waterfront area for live race updates. Each tower used four QSC Model 3800's and four Model 3350 amplifiers.

In addition to the towers, a stage was built for a live performance by Steve Smith's band, Vital Information. The stage area used twelve QSC Model 3800's and eight Model 3350's. Stage monitors were powered by QSC MX 2000 and 3350 amplifiers.

Globe Arena in Stockholm Powered by QSC

The sound system for the 16,000 seat Globe Arena currently under construction in Stockholm will contain 86 QSC power amplifiers. For the main area, 40 QSC Model MX 1500 will be used to drive a central cluster loudspeaker system. An additional 46 QSC Model 1200 amps will be used to power speakers to surrounding public areas. Acoustics for the job are being handled by Lennart Thunemalm AB. Installation of the QSC equipment was by Englund Musik of Sweden.

JBL Pro Installed at Alexander Brest Planetarium

JBL Professional has a new prestige installation at the Alexander Brest Planetarium in Jacksonville, FLA. In addition to its JBL Professional sound system, the Planetarium contains hundreds of special effects projectors, an intricate laser system and 8-70 mm films.

The JBL sound system at the Alexander Brest Planetarium includes 24 sub woofers at seat level and six custom cabinets with 11 speakers in each behind the dome. The six discrete channels with one sub woofer channel combine to total 18,000 watts of R.M.S. power. The system's flat power response is 20 hertz to 16,000 hertz, with a full power lift of 20 dB below 100 hertz. The maximum Sound Pressure Level (SPL) in the chamber is 128 dB linear. The planetaruim's sound system was designed, constructed and installed by Florida Sound Engineering Company of Jacksonville, FLA.

Panasonic/Ramsa Installations

Panasonic/Ramsa has a new installation at the Arco Arena in Sacramento, CA. The installation included a Ramsa sound system, a Panasonic Astro Vision scoreboard, a Panasonic CC- TV system, Industrial video equipment, Panasonic point-of-sale and Panasonic telephone systems. The Ramsa sound system was installed by Associated Sound in Sacramento, CA and Acro Media in Culver City, CA and was coordinated by Pixel of Albany, NY. The system was designed by Howard Smith of Smith-Fause, Los Angeles, CA.

Another Panasonic/Ramsa installation took place in the Winter Garden, a large cathedral room in the World Financial Center building in New York City. A special distibutive system was designed by Howard Smith of Smith-Fause to tackle a room defined as "acoustically difficult" by Panasonic/Ramsa marketing manager Chris Foreman. The contractor for the Winter Garden system was Peirce-Phelps of Philadelphia in cooperation with Robert Drake and Associates in NYC.

CRC Contractors for Tower Records

CRC, general contractors of Citrus Heights, CA has just completed the installation of the audio and video systems in the new Tower Records stores in Stockton, CA and Carle Place, NY. The company will be installing systems for Tower Records in the Stonestown Mall in San Francisco in November, in Tyson's Corner, VA in December. CRC has installed systems in almost all the Tower Record stores in the US and in England.

The system in the Tower Records in Stockton included 20 JBL-120 TI speakers, 13 KXM 200 Sony monitors, JBL power amps, Proton Pre amps, a Technics SL1200 turntable and a Technics CD player.

Crest for the Celebrity Theater

The Celebrity Theater in Phoenix, AZ has updated their main sound system with Crest Amplifiers and EAW loudspeakers. Tony VanNote of Advanced Audio Systems in Phoenix, installed nine Crest 4001's and three Crest 8001's. EAW KF600 and SB250 loudspeaker systems were also installed.

CALENDAR

OCTOBER 3–4
Kentuckiana Sound Seminars
Indianapolis, IN
Andy Baker & Associates (317) 253-9667

OCTOBER 6–8
ICIA video, audio-visual computer seminars
Atlanta, GA
Debbie Hafer (703) 273-7200

OCTOBER 12–15
IBMA
Fort Lauderdale, FL

OCTOBER 19–21
Network 90's Telecommunications Conference and Expo sponsored by USTA and USTSA
San Francisco, CA
Paul Roguski (202) 835-3158

OCTOBER 23–25
NECA
New Orleans, LA

NOVEMBER 3–6
AES
Los Angeles, CA
(212) 661-8528

NOVEMBER 14–18
ASA
Honolulu, HI

NOVEMBER 18–20
LDI-Lighting Dimension International '88
Dallas, TX
Patricia MacKay (212) 677-5997

NOVEMBER 10–14
NCAC
Honolulu, HI
(201) 379-1100

NOVEMBER 29 — DECEMBER 1
Unicom 2 Expo and Conference
Sponsored by NATA
Dallas, TX
Kay Hynson (703) 273-7200

JANUARY 15–18 '89
PTC '89 Pacific Barber Telecommunications Council
Honolulu, HI
Richard Barber (808) 941-3789

FEBRUARY 2–4
INFOCOMM sponsored by ICIA and AECT
Dallas, TX
Kay Hynson (703) 273-7200

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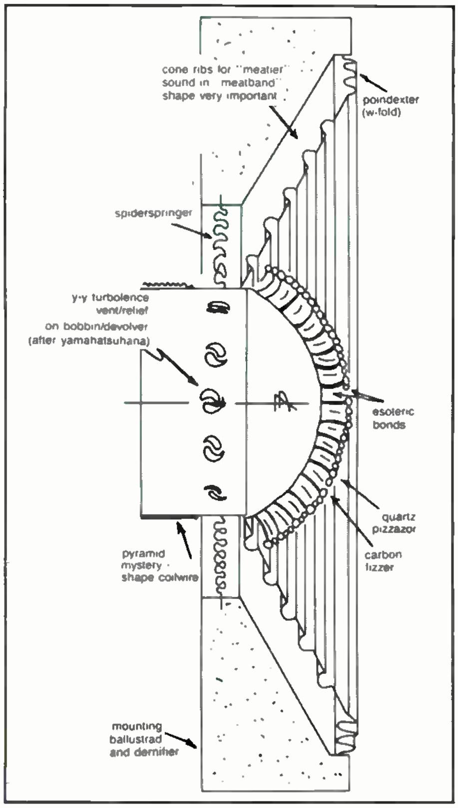

Figure 4. Diaphragm Assembly

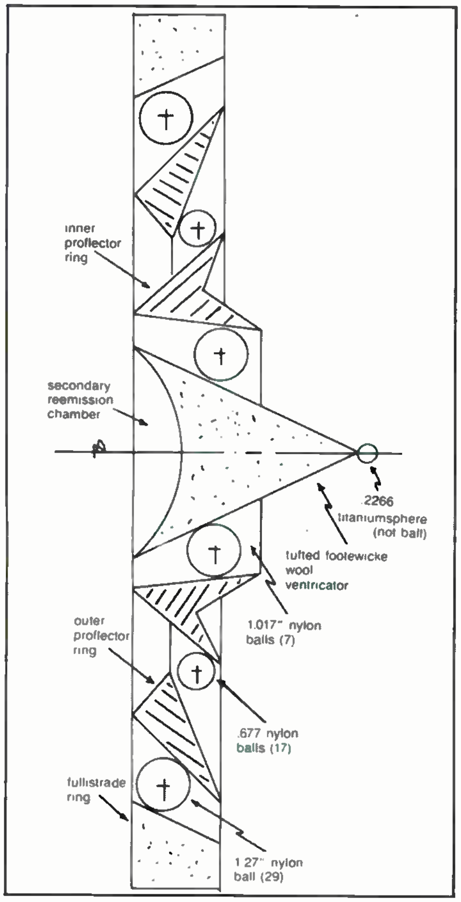

Figure 5. Deflector Assembly

This will give you tone response. If it isn't flat from 20 to 20,000 cycles, rebuild all the drivers and start over. Don't give up.

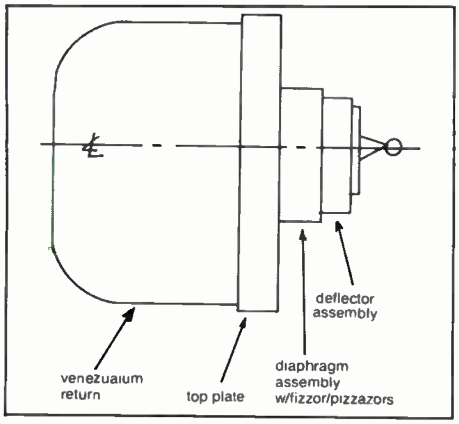

Figure 6. Driver Assembly

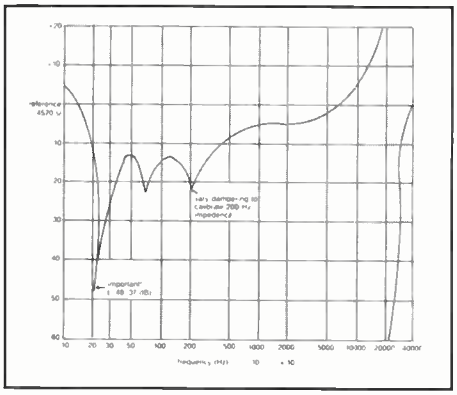

Figure 7. Impedance Calibration

There: I've already told you too much. Except, of course, how to build a correct listening room. What's that? You don't have one? Shame, shame. All along, I assumed. . .ah well. I guess it's time once again to bite the bullet, and reveal the next stone tablet. ■

Next Month: "Building Your Own Dr. Wokka Calibrated Listening Room."

LAB TEST

(continued from page 59)

tainly was localized well within the unit—a plus for intelligibility, but what is the metric for this? Some "time smear" (or so-called "phasiness") is audible through the crossover region. This is mostly attributable to the cone breakup of the 12-inch. This phenomena is found in all loudspeakers, and to what degree this phenomena is found is considered when designing the transducer itself, and the systems in which it is to be used. Factors of cone shape, stiffness, and mass are all balanced to output and linearity. While the mid-range "phase response" does not sound like your favorite pair of $5,000 studio monitors, they certainly sound monitor-like for a high-powered/high-efficiency sound reinforcement loudspeaker.

The PAS 12-inch has a sensitivity of 101 dB re: 1-watt/1-meter. This high-range sensitivity comes at the price of some cone breakup (not unexpected when a light cone is driven hard). When listening to such a speaker at moderate levels in a low-reverberant space, this breakup can be heard and measured. If measured, cone breakup can be misinterpreted in time-domain analysis. Therefore, measurements in this domain must be carefully made, as well as appropriately analyzed and described. What we hear and perceive does not always agree with a simple single-point test mic not connected to a brain.

Conclusion

The unit has been successfully used on stage as a high-powered, high-quality, low-profile monitor. My experience with the PAS, mostly in Atlantic City casino showrooms and clubs, has demonstrated that it requires very little equalization for feedback elimination in these "every dB counts" situations. Similarly, in PA applications where either a single small source, a simple array, or distributed sources are needed, the PAS is a prime candidate. The unit is rugged, sounds good, has a good deal of acoustic output relative to its size, and needs very little "tweaking" to make it work. It is important to be able to predict how the unit will behave in a given situation, and this ability comes from a simple observation of the physics (read physical dimensions) of the unit, observation of the polar patterns, and the intended application/environment. ■

Klapholtz is technical editor of Sound & Communications.

Measurements

by Farrel M. Becker

The Professional Audio Systems (PAS) model PI-122P is a two-way, passive, coaxial loudspeaker system. Contained in a ported trapezoidal-shaped enclosure, the system consists of a PAS CX-1280C 12-inch coaxial loudspeaker and integral "TOC" crossover (TOC stands for "Time Offset Correction," and is Professional Audio System's term for electronically aligning the acoustic centers of loudspeakers). The high frequency section of the loudspeaker uses a compression driver and a high frequency horn that, according to PAS, is of the constant coverage type.

The magnitude of impedance of the PI-122P is shown in figure 1. The minimum impedance of 4.6 ohms occurs at 11.8 kHz. The peak at 38 Hz is due to the port and is 52.3 ohms. The impedance at the woofer's resonant frequency (113 Hz) is 19.1 ohms.

Figure 2 shows the on-axis ETC (Energy Time Curve). The loudspeaker emits significant energy over a 1.3 millisecond period.

The anechoic, on-axis frequency response is shown in figure 3. The 1 watt/four foot sensitivity at 1 kHz is 94.7 dB. The EIA rating is 47.2 dB and the 1 watt/1 meter rating is 96.4 dB. PAS quotes the loudspeaker's 1 watt/1 meter sensitivity at 101 dB. Figure 4 shows the results of near-field low frequency measurements. The response of the woofer and the port are shown individually. The curve marked SUM is the resulting response when the power sum of the two individual curves is taken. This is representative of the far-field response. The 3 dB down-points are 60 Hz and 12.4 kHz. PAS quotes the 3 dB down-points of the PI-122P as being 60 Hz and 19 kHz.

PAS quotes the phase response of this loudspeaker as being plus or minus 10 degrees from 100 Hz to 10 kHz. Figure 5 shows the phase response of the PI-122P from 0 to 20 kHz. For figure 5, the analyzer was "tuned" for the mid-range. Figure 6 shows the phase response from 0 to 3 kHz and was tuned 160 microseconds later for that range. Here the crossover point of 1.2 kHz is clearly evident. Notice the interference notch at the crossover frequency in figure 3. The phase of the tweeter is shifted approximately 90 degrees relative to the woofer. The phase response is not quite plus or minus 10 degrees, but that is a hard spec to meet with any loudspeaker.

Figures 7 and 8 show the ETCs for 30 degrees off-axis horizontally and vertically respectively. The time response off-axis is quite similar to that on-axis. This is one of the characteristics of coaxial loudspeakers.

The horizontal and vertical polar plots of the PI-122P are shown in figures 9 and 10 respectively. These are composites of the frequencies in the 2 kHz octave band. At 2 kHz the coverage angles are 70 degrees horizontally and 100 degrees vertically. The vertical pattern begins to beam above 7 kHz. The horn's larger horizontal dimensions prevent beaming in the horizontal plane until 15 kHz.

At 200 watts the PAS PI-122P will provide 100 dB of direct sound at 30 feet. It should be ideal for a small club or to provide foreground music in a restaurant. Its built-in rigging points will make installation a snap and its appearance right out of the box is suitable for public viewing. (Paint or stain it if you like, but I think it looks good just as it is.) ■

Becker is an acoustical consultant with Audio Artistry, Kensing, MD.

AD INDEX

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The Words of Technology

Punch! Slam! Bang! Sound like the Olympic boxing competition melee? Perhaps, but to the silent majority of audio, these terms reflect today's high-tech, musically driven culture. Mass media brings us changes in style and music faster than the English language can adapt. This expanding gap is affirmed by troubled dictionary publishers on all fronts. "Sweet," "velvet," and "pristine" may sound like words straight out of a paperback novel, but these are descriptors of sound quality commonly used in audio. They are the antithesis of "gritty," and "biting"—terms used routinely in the personification of the first two generations of solid-state amplification.

"Through a dark, murky heaviness...," sounds like a narration by Vincent Price. There are innumerable subjective characteristics ignored by our body of science—enough to fill this whole page. With the efficiency of superconductivity, science marches on at the speed of light, at times blissfully unaware of or insensitive to the creativity and illusions of life itself. As we design more hardware, and as we use more digital audio, we move closer to scientific sterility and farther from human subjectivity.

Along with the vocabulary we've developed to describe our subjective responses, are the terms we use to describe quality. "Modern," "professional," and "high-quality" simply don't have the same impact or meaning they used to. The latest term to fall by the wayside is "high-tech." Our language of electroacoustics has changed from the days when *loud speakers* were used for *re enforcement*. For example, take the word "sonorous" (no, not the underwater distance-ranging system), used to describe a sound that is characteristically deep or rich. "Sonorous," no longer used, was a common element in the vocabulary of acoustics and sound in the early days.

A client and I were discussing some acoustics problems recently, concerning his church. The man, an engineer himself, was able to describe explicitly the geometry of the room, its architectural finishes, and the positions of the various sources and listeners. From the description of the curve in the ceiling, with its thin ceiling tile covering, it seemed as if there was a midrange echo problem. I asked him if in fact there was a high-frequency echo, in order to verify my suspicions. He honestly said that he didn't know what that meant. I asked him if there were any *zingy* sounds. He enthusiastically exclaimed, "Oh no, not at all!"

This year, at the 85th Convention of the Audio Engineering Society, we will be gathering a large group of specialized disciplines within our field. These will include the fields of recording and sampling studios, broadcast, music, consumer electronics, sound reinforcement, high-level sound, commercial sound, disco, theatrical, religious sound, academia, and the specialized areas of microphones, loudspeakers, analog signal processing, amplification, and digital audio—just to name a few. Los Angeles will certainly have a diversity of special interests, cultures, and languages.

While we all look forward to attending the papers presentations and workshops, and to visiting the booths, some of the most enlightening moments at these gatherings are the interactions between our colleagues; between the musicians, engineers, practitioners, professors, and students there certainly is a dichotomy of vocabulary. "Heavy-to-light," "hard-to-soft" may not be easily quantifiable in our mathematical models or measurements. However, they may be more of a topic as we advance in our power to process and analyze in this *modern*, *high-quality*, *professional digital-audio* convention.

Jesse Klapholz
Technical Editor

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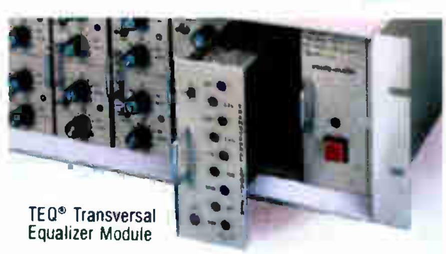

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