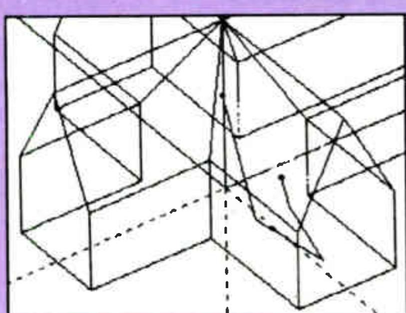
SOUND & COMMUNICATIONS

Volume 35 Number 9

October 18, 1989

SPEECH INTELLIGIBILITY

Can speech intelligibility actually be predicted, or is it a wayward science based on trial and error? A lot of software programs are making valiant efforts in predicting levels of intelligibility. How good are they? Mike Klasco reports. **32**



Expanding Into New Markets

How does the sound contractor expand into new markets — and should he? *Sound & Communications* presents a seminar on tangential markets that can make money, and how to make money with current endeavors. Whether it's dbs, residential work, wire installation, hospital communications or any other expansion, the bottom line is the uppermost concern. **12**

AES CONVENTION

It's a trip even if you live in the convention city. This year's 87th Audio Engineering Society convention gives a hint of the nineties and wraps up some of the technical breakthroughs of the eighties. What will you find at the New York Hilton on October 18? Turn these pages. **18**

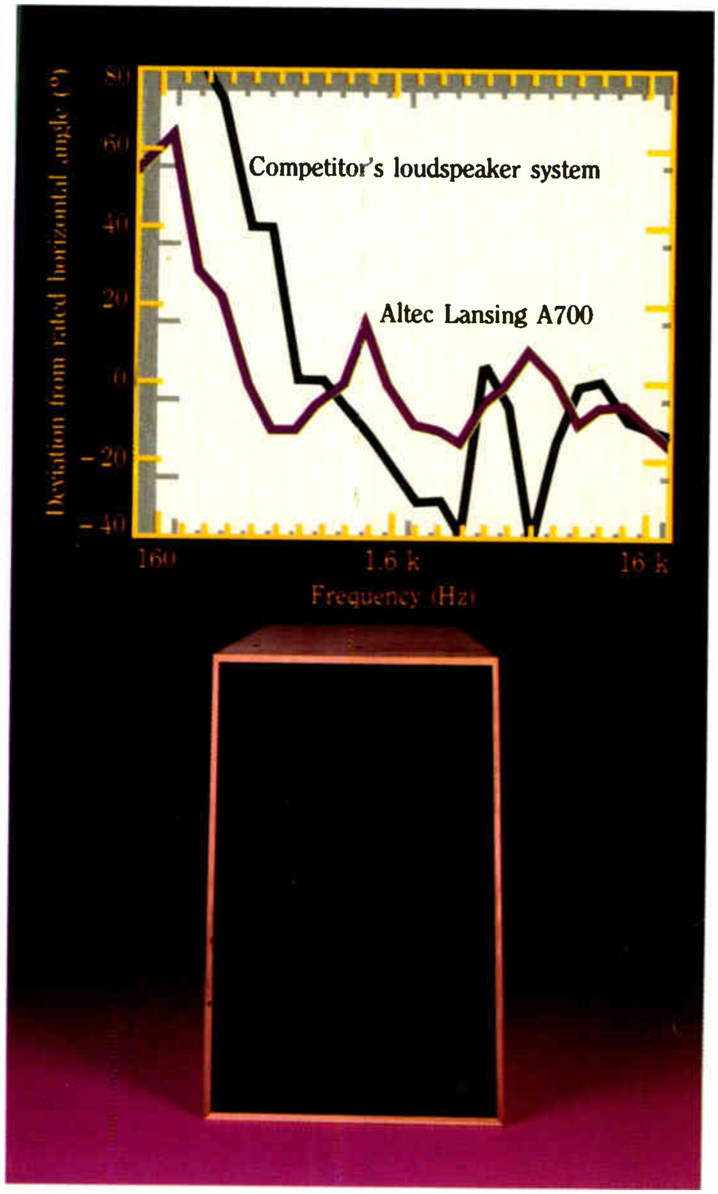
STADIUM SOUND

Steinbrenner and signal-to-noise ratio may be the new amazing duo. Across the country, from coast to coast, sports and sound go together. This month, *Sound & Communications* looks at the latest upgrades and installations in stadiums. Who's doing what and what's being used? In the mino
business being do
Yankee Stadium,
Arena, River Dow
dome, Tucson Toro

In loudspeaker systems, uniform coverage equals superior response. The frequency response of a loudspeaker is the acoustic pressure level it will produce with a constant voltage input, over a variety of frequencies. The coverage of a loudspeaker is the angular wedge of sound radiation it produces; that is, the area it will serve in actual use. What have frequency response and angular coverage to do with each other?

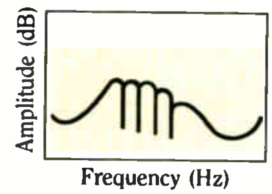
It isn't very difficult today to create a loudspeaker with excellent frequency response directly on-axis. This axial response may actually be one of the less interesting ways to measure response. To know the frequency response most audience members will hear, we must consider the coverage of the loudspeaker. This is especially important in the case of multiple loudspeakers used jointly to cover adjacent areas.

If a loudspeaker coverage pattern narrows down in some frequency bands, listeners outside this diminished coverage hear it as a response deficit: a dip in the response at those frequencies. If a loudspeaker coverage pattern widens at other frequencies, and this loudspeaker is used in conjunction with other similar loudspeakers for wider-area coverage, listeners in the overlap area hear sin x/x interference, sometimes popularly called comb filtering.



The graph shows deviation from rated horizontal coverage angle, for the Altec Lansing model A700, and for a popular loudspeaker system of approximately the same size and price.

The coverage of both loudspeakers widens at low frequencies, however the A700 exhibits much greater uniformity over this range. Note also the A700s smoother behavior in the critical upper octaves while the competitor's unit allows the pattern to collapse at two frequencies.



Coverage overlap produces interference; reduced coverage produces frequency response gaps.

The designer of a loudspeaker system must take great care that its coverage remains constant over most of its useful frequencies. This way, most listeners in a correctly-designed application benefit from the best frequency response the loudspeaker can produce. Gaps or overlaps in coverage all deteriorate the frequency response the audience will hear.

What is the benefit of uniform loudspeaker coverage? Superior frequency response for all listeners.

Fact.

In Loudspeaker Systems, Uniform Coverage = Superior Response

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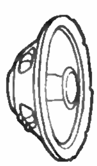
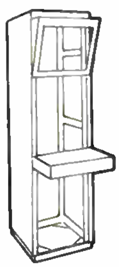
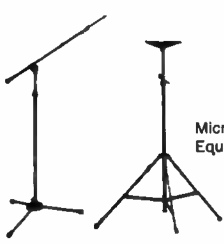


Professional Sound Loudspeakers

Voice/Tone Loudspeakers



Cabinets, Racks and Consoles



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TECHNOLOGY UPDATE

In the past few years, while the format wars raged in large scale digital recording, and the small R-DAT format was engaged in a political limbo, there was a noticeable quiet over mention of another format developed about the same time as R-DAT — S-DAT. Now that digital tape recording is mainstream for music, DASH and PD coexisting, and the political maelstrom has subsided over R-DAT, our attentions turn to just what is happening with S-DAT.

We talked to Don Morris at Yamaha about S-DAT and its possibilities.

S-DAT of course stands for stationary head digital audio tape, and as such can be used to define DASH, Pro Digi, or even, as Morris says, stretched to include a reel to reel machine. (However, recording is necessary. "With R-DAT you always work with a backup," says Morris. The problem with rerecording is that if there's a brown out, or "some assistant presses 'pause,' it's all lost."

Error correction coding for S-DAT is not consistent with other formats. But, Morris says, there's no consistency from manufacturer to manufacturer anyway. S-DAT will probably have proprietary error correction.

How small is S-DAT? It can use 8mm tape. How many tracks? Certainly more than two. The video applications for a small S-DAT machine are the same as any small multitrack. It's good for building audio tracks in post, on-line and off-line. The smaller format lends itself to video because video doesn't "need megatracks as in pop music."

S-DAT BASIC PARAMETERS

(these are the *basic* parameters of S-DAT, with which several companies are working. Working, marketable models may or may not in the end subscribe to these standards.)

Cassette Size (mm)	Tape speed (mm/s)	Writing speed (cm/s)	Transmission rate (MBPS)
86 x 55.5 x 9.5	47.6 43.7	4.76 4.37	2.4 12.2

S-DAT, when referred to by the Japanese engineers we've met who were working on it, has been generally thought of as small format with much greater professional applications than R-DAT.)

Morris amplifies: Since stationary heads on the S-DAT machine stay put, head alignment is not a problem. But the main practicality is in punching in, or dealing with material on tracks one and two, for instance. In rotary DAT, when recording tracks three and four, one has to rerecord tracks one and two so that they sync. S-DAT is more congenial to multitrack, since no re- And "video wants safety," which R-DAT doesn't provide. It is no harder, according to Morris, to interface S-DAT than to interface any other format.

What are the applications for S-DAT in background music, for example? It would probalby have to be looped. The length of recording time on S-DAT is shorter, since the stationary head requires increased speed of tape travel.

Yamaha is, according to Morris, investigating all recording formats — disk based, R-DAT, hard disk, write once. And may be ready to move in one — "or many" — directions. Stay tuned.

SOUND & COMMUNICATIONS

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And it offers you the kinds of features that theirs doesn't . . .

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C O N T E N T S

Volume 35 Number 10 October 18, 1989


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FEATURES

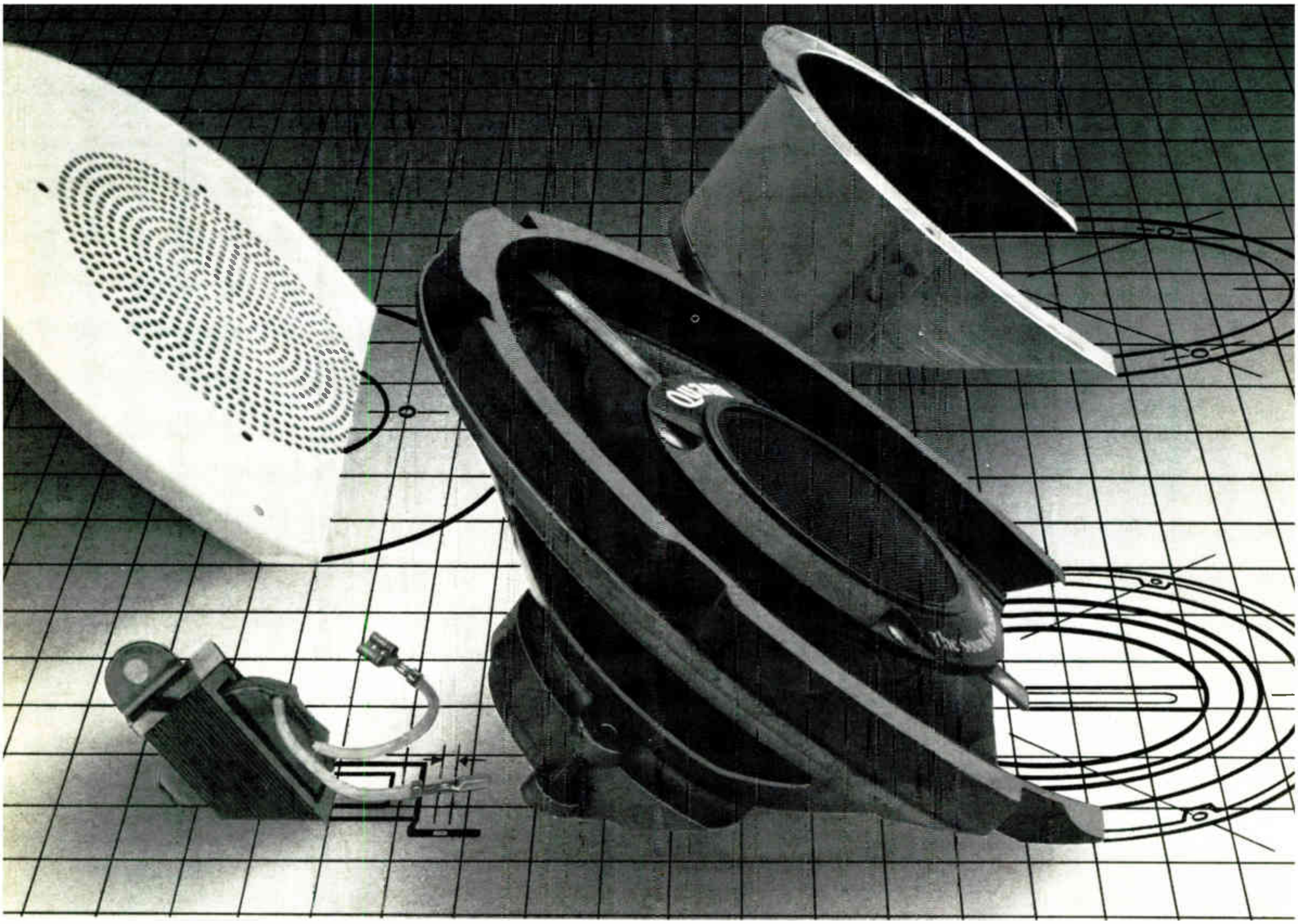
COLUMNS

DEPARTMENTS

SOUND COMMUNICATIONS

Coming in November. . .

An examination of the drive-through communications market. Also: assistive listening systems — new politics could spell opportunity for contractors.



If you take us apart, you'll take us on.

When we decided to expand the Quam line to include ceiling baffles, backbox enclosures and assemblies, we wanted to do more than just complement our loudspeaker offerings. We wanted to give the contractor another choice. Judging from the growing list of contractors who have switched to Quam, we did just that, with an unbeatable combination of quality, price and service.

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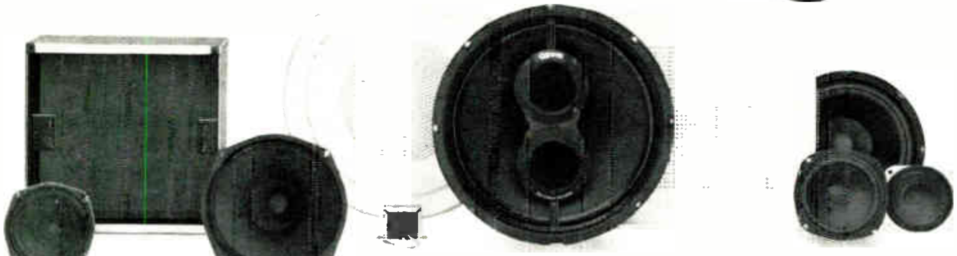
Quam: The Sound Decision



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NEWSLETTER

AES NEWSMAKERS

What products should you expect to see at the AES convention? At press time few companies had made final plans, but some of the things to look for are:

Sennheiser is showing the S-7M high powered Infrared Theatre Listening System and the MKH 50-P48 condenser microphone, a new member of the symmetrical transducer, transformerless MKH family... Audio Processing Technology, a subsidiary of Solid State Logic, will demonstrate the apt-X 100 digital audio compression system... Bruel & Kjaer is launching the Type 4012 prepolarized condenser microphone... Ultra Analog is supplying the original CTI/dbx 20 bit A/D converter IC set...TAC is showing six versions of their Bullet line of consoles... Oxmoor is introducing two new distribution amplifiers....

Milt McNally, well known from his involvement at a/d/s and at Carver, is now with Audio Animation who will be showing their Muse digital mastering console.

AMS is showing its ST250 stereo microphone and control unit, switchable between stereo and mono... Soundtracs is showing four new products, including the 40-channel version of the SPA sound reinforcement console... Agfa will have a slide presentation on the Agfa XT tape restoration process, at its booth... Telex is showing wireless headsets... Harrison Systems (GLW) has new automation for its Series 10 console... Studer is expected to show a 48-track digital unit...

A.R.T. is introducing an equalizer and a new multiple effects processor.... GML is showing a new limiter and several "automation" devices....Intersonics will be showing the Contra Bass servo-motor subwoofer... Tannoy is showing the NFM-8 (DMT) near field monitor, and will have drawings for studio reference monitors... API Audio Products is introducing an all discrete in-line recording console with the GML Series 2000 automation environment. (Sound & Communications will present a full wrap-up of the AES show in an upcoming issue.)

AUDIO COMMUNICATIONS ANNOUNCEMENTS

Robert W. Peters has been appointed general sales manager of Audio Communications Corporation's Business Music Division. The Houston-based Audio Communications has finalized its exclusive dealership arrangement with Audio Environments Music network (Seattle, WA) to distribute all direct broadcast satellite, FM direct, and on-premises tape programming for the Houston market.

Audio Communications was also appointed the national marketing entity of Telemusic USA, also based in Houston. Telemusic is a new music and message-on-hold company.

LICENSE AWARDED

Sound Control Technologies, Inc. of South Norwalk, Connecticut, has been granted exclusive license by Jaffe Acoustics, Inc. to the latter's newly issued Patent #4837829, "Acoustic Sound System For A Room." SCT has incorporated the electronic and acoustic principals covered by this patent (and the previously licensed Jaffe Acoustics Inc. Patent #3992586, "Boardroom Sound Reinforcement Systems") in its Voicelift voice reinforcement system and its Teleplex teleconferencing systems. The newly issued patent broadens the adaptability of SCT's systems to include boardrooms, conference rooms, and training areas.

NEWSLETTER

APOGEE APPOINTED BY ITALIAN MANUFACTURER

Apogee Electronics Corp of Santa Monica, California, has been appointed the exclusive US distributor by Audioscope, an Italian pro-audio manufacturer and developer of the Model 9000 audio measurement system. The Model 9000 consists of a 3U rack-mount card frame and external color video monitor. Its modular facilities are suggested by Apogee for applications from live sound studio to monitoring.

PONCHER HEADS EDS BOARD

Charles Poncher, president of Hawk Electronics, Inc. (Wheeling, IL) has been elected president of the Electronic Industry Show Corporation. Poncher, also president of the National Electronic Distributors Association, will head the planning process for the 1990 Electronic Distribution Show and Conference. EDS '90 will be held at the Las Vegas Hilton April 24-26 of next year.

Other officers elected to serve with Poncher during the annual meeting of the EISC board of directors were David McCoy of Cartwright & Bean (Atlanta, GA) as vice president, and Blair Haas of Bud Industries (Willoughby, OH) as secretary/treasurer. David L. Fisher continues as executive vice president.

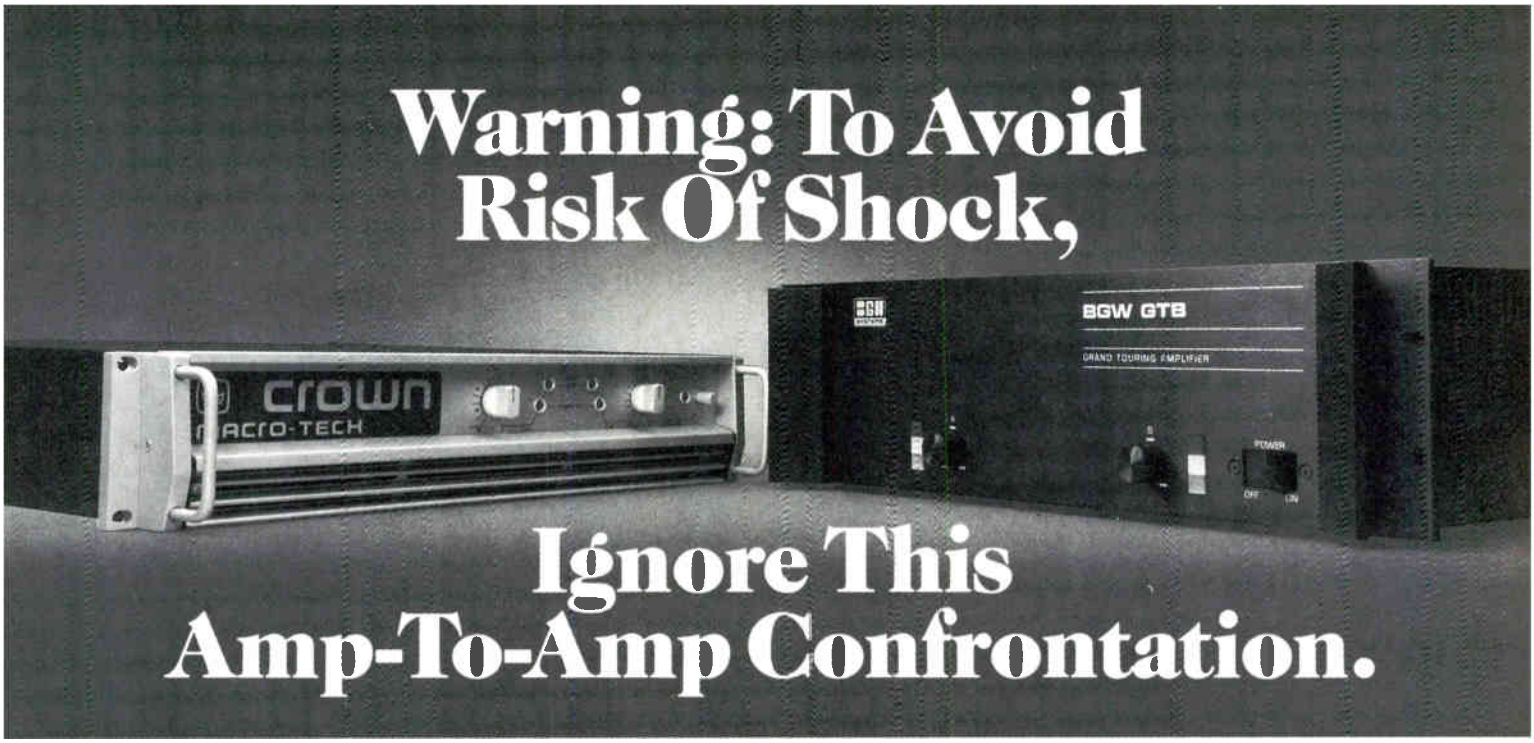
EDS is operated not-for-profit under the sponsorship of the Electronic Industries Association, the National Electronic Distributors Association, and the Electronics Representatives Association. Directorships and elective offices are shared among the three groups.

UNIVERSITY, GRAYBAR SIGN DISTRIBUTION AGREEMENT

University Sound, Inc. (Sylmar, CA) has signed an agreement with Graybar Electric Company for the St. Louis-based firm to distribute University's commercial sound products in the US. "We are very pleased to announce the newly extablished relationship with University," said Doug Peck, Graybar's national telecommunications marketing manager. "We anticipate the agreement will allow both Graybar and University to realize growth in existing as well as new markets."

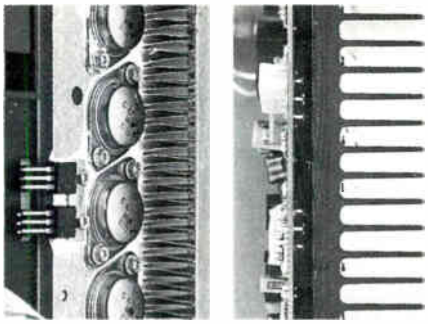
RAULAND-BORG, SIMPLEX SETTLE LITIGATION

Rauland-Borg Corporation (Skokie, IL) recently announced it has reached a settlement with Simplex Time Recorder of Gardner, Massachusetts, that dismisses all claims and counterclaims. Rauland-Borg granted to Simplex a license under its US and Canadian patents covering microprocessor-based, user-programmable, electronic master clocks. The lawsuit was initiated after an impasse had been reached between Rauland-Borg and Simplex wherein Rauland-Borg offered to include Simplex in its licensing program for the patent on its Model 2424 User-Programmable Clock. Simplex originally infringed the patent based on that product.



Let's be frank. We're out to change your idea of what—and who—makes a professional power amplifier. So if you just bought a Crown MacroTech, turn the page — this comparison won't be a polite one. But it will stick to the facts.

A look inside these two amps will give you a better idea of why BGW amps like the GTB Grand Touring Amplifier are built like no others in the world. And raise some questions about Crown MacroTechs.



Left: *The MacroTech uses mostly air to dissipate heat, not metal. The closely spaced fins are vulnerable to airborne dust and dirt.*

Right: *BGW uses ten pounds of aluminum to absorb thermal transients, extending power transistor life.*

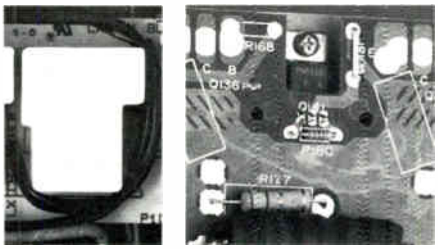
TAKING THE HEAT

If the MacroTech heat exchanger reminds you of an air conditioner, you've grasped its design. This approach works, at least until dust and dirt clog the fins. But as soon as the air flow slows or stops, temperature rises. Soon after that, the Crown shuts off — it could even fail.

The GTB uses massive extruded aluminum heat sinks with widely spaced fins. The mass of metal absorbs thermal transients without straining the fan. And without quick changes in transistor temperature. That's important: Transient musical loads put the worst kind of stress on power transistors. The effects of thermal cycling fatigue may not show up until after the warranty, but they can destroy lesser amps. Meanwhile, BGWs keep right on delivering clean, reliable power.

REAL SPEAKER PROTECTION

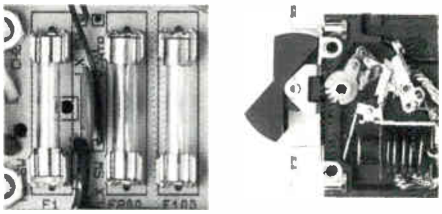
Most amps today are direct coupled, so a blown output transistor (the most common failure) connects the power supply directly to the speakers. Earlier MacroTechs had no protection against DC. Now Crown has learned their lesson — or have they? The sensing circuit and relay they now use shuts off the power transformer, but allows the filter capacitors to discharge stored DC energy directly into your drivers — risking real damage.



Left: *Crown uses a slow-acting, less reliable relay. It can allow the filter capacitors to discharge stored energy directly into your drivers.*

Right: *BGW's modular power output section protects your speakers against DC damage with an instantaneous Thyristor Crow Bar. And the module is easily replaced in the unlikely event of failure.*

BGW pioneered DC speaker protection in 1971. We stopped using relays years ago, when they no longer met our reliability standards for BGW amps. The GTB, like all BGWs over 200 Watts, uses solid-state Thyristor Crow Bars to keep DC from ever reaching your valuable speaker cones or compression drivers.



Left: *Time is money, and with Crown's MacroTech you can lose plenty of both: You have to pull it out of the rack every time a fuse blows.*

Right: *The GTB's power switch is also a rocker-actuated magnetic circuit breaker. You can reset it in a second if power lines hiccup.*

MAKE YOUR OWN COMPARISON

Before you buy or spec your next power amp, call us at **800-468-AMPS** (213-973-8090 in CA). We'll send you tech info on BGW amps and the name of your nearest dealer: He can arrange a demo of any BGW model against any amp you choose. Then you'll be able to appreciate the advantages of BGW engineering with your ears, as well as your eyes.

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Expanding Into New Markets

At the National Sound and Communications Expo in May, *Sound & Communications* magazine sponsored a seminar on "New Markets for Sound Contractors". Participants were Tim Newcomb of Newcomb Electronics, Chris Stevens from Audio Access; Bob Toerner of American Sound; Bob Reim from Acromedia; and Bill Ray of William Ray Associates. The moderator was Judith Morrison of *Sound & Communications*.

MORRISON: As an editor at Testa Communications, I've supervised a lot of the copy that goes into the magazines that we publish. *Sound & Communications* is one of those magazines. We also get involved in post-production of audio, video and film; in M.I. sales; in various aspects of the sound and communications industry not always directly related to contractors. We also produce convention television. At this convention, we do NSCA-TV News. That's how *we* cover this market. In our travels, we obviously go to a lot of conventions, as I'm sure a lot of you do. Last week, I was in Las Vegas at the Broadcasters Convention and this week I'm here at the Sound and Communications convention. Next week is the CES Convention in Chicago. Right after that is the NAMM show. And if I chose, I could go to the Interior Designers Convention in Toronto. Last week, I could have gone to the AES Digital Audio Forum in Toronto. I could go to conventions all year round. So could you.

There are lots of things to learn. Lots of things to see. But the thing that always amazes me is no matter which city I am in or what convention I'm at, I'm astonished at how many friends I've seen: People who are crossing over from one industry to another.

This Sound and Communications convention is growing, growing, growing, and I think it is because there are roots in the sound contracting community that are not going to be lost. But there is an expansion and a contraction going on. There are musical instrument retailers who are very interested in getting into church sound. It's a way for them to expand their market. There are sound contractors getting into residential work, but there are also hi-fi specialists moving into sound contracting. It's not easy for them sometimes. They don't have the labor and they don't have the design facilities. But they're doing it.

So, I have gathered here an illustrious group of panelists to talk a little about expanding into new markets or watching for who is expanding into your market. To my far right is Tim Newcomb, from Newcomb Electronics, who has expanded into several markets. Chris Stevens is with Audio Access and was the founder of Phoenix Systems, a custom installing contractor. Bob Toerner from American Sound is into a lot of different industries. Bob Reim is president of Acromedia. And Bill Ray, who heads up William Ray Associates, is a rep with a lot of illustrious lines: JBL, Otari, Audio-Technica, dbx. These gentlemen are going to talk a little bit about what they see as new markets for people to go into. Bob, do you want to respond to that?


Left to right: Tim Newcomb, Chris Stevens, Bob Toerner.

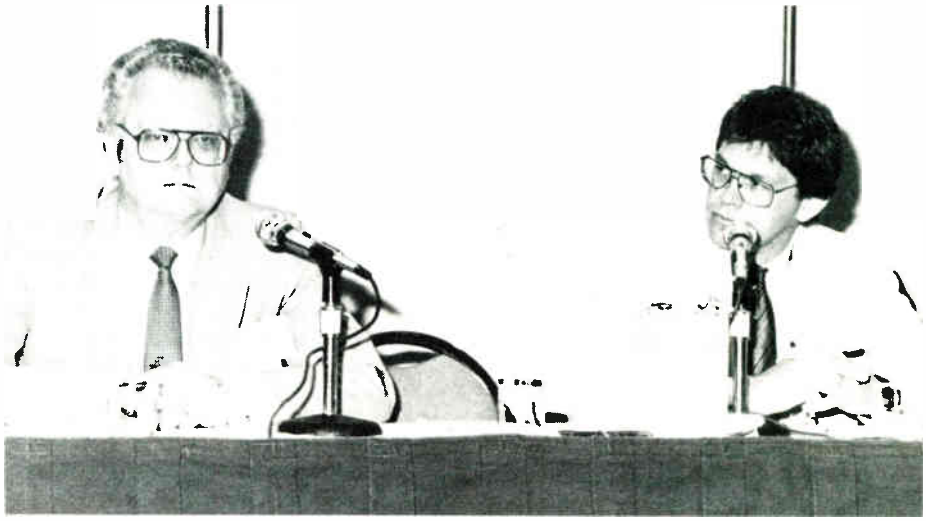
TOERNER: My dad started American Sound 43 years ago and laid a good basis for expansion in that he didn't rule out anything that was close to the sound business and that he could build on and expand on. Now, after 43 years, American Sound has eight sales engineers. We work out of a Cincinnati office; we also have offices in Dayton and Lexington; and we now cover things like commercial sound systems that have expanded to cover standard paging, churches, restaurants, sports bars, background music. We're now taking a good strong look at direct broadcast satellite.

The other areas where we have

specific sales engineers are in hospital communications, professional video systems — and that includes CCTV and security telephones. We have one man now who specializes in wire sales. That person is taking a long look also in a new area of card access and fire alarm systems. Video projection and boardroom communications systems are starting to go into their own as a separate category. Land systems, also MATV systems, some of you may have already dabbled in. Not to mention the ever popular cellular phones as possible edge to what you're doing now. So, we're kind of broad based. We're always looking for something new as long as it's profitable.

MORRISON: That's the key word — profitable. Tim, you've gone another route to an extent and gone residential.

NEWCOMB: Right. We've actually gone two routes. But the residential route is actually an outgrowth of our sound contracting business. We were doing large sound systems, for example distribution facilities and manufacturing plants. And the owners were coming to us and saying, "Gee, I have a stereo at home that some guy from New York installed for me. I don't have the slightest idea how to turn this thing on. Come on over to my house and check it out." So, we started to use some of the techniques we've always used in commercial sound contracting, which is to keep the systems very simple. And we developed a sound system for a particular client which is very easy for him to operate. It featured a remote control in every room, and we expanded on that to have complete systems and have sold quite a few of them across the country. We've ended up doing projects in Boston, New York, Palm Beach, Aspen. It's been a very interesting and stimulating business. And it's the kind of business where you have a very upscale consumer and price isn't the only delineating feature. So you're in a nice area to make a good profit if you come through with the goods.



Left to right: Bob Reim, Bill Ray.

As for other departures from traditional sound contracting: Computer systems use very similar cabling to audio systems. The computer guys always think their systems are pretty hi-tech and their cabling demands are pretty rigid. I tell you — if you can install audio cabling, you can install computer cabling. It's been a very good source of revenue for us. The funniest comment we come across is when a computer vendor or a computer user comes in to look at the wiring up close and sees everything nice and neat and labeled, and they can find everything, they say, "Gee, we never thought it was supposed to look like this." So we get tremendous credibility. There are a lot of hacks out there, and there's a way to go in and really set yourself apart from the rest of the crowd and make a good profit.

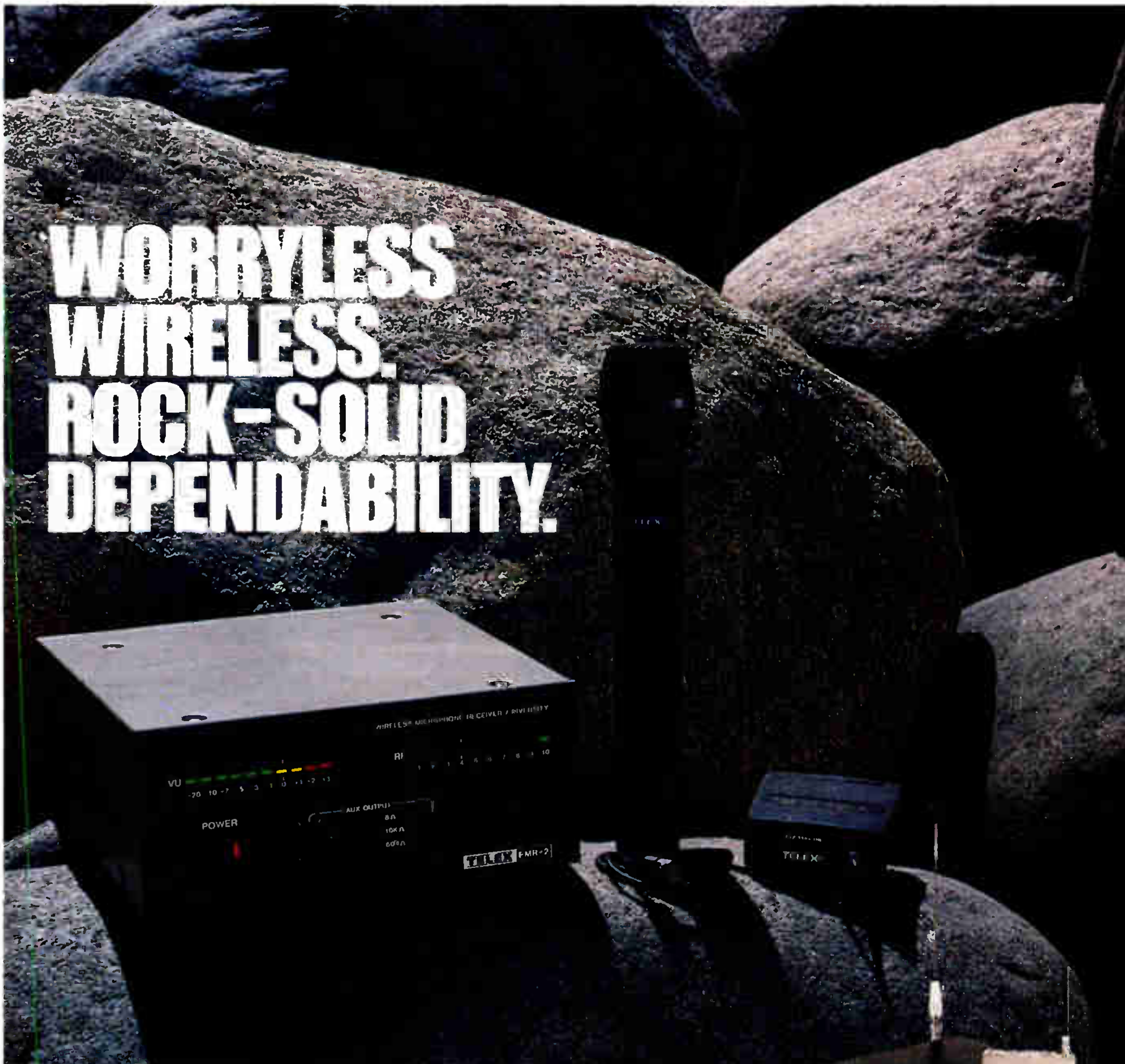
MORRISON: Make a good profit and neaten up those cables. Chris, you were in commercial, you got out of commercial. Now you're back here.

STEVENS: About 15 years ago when we started Phoenix Systems, and at this point it would have been in the neighborhood of $150,000-$200,000 a year, we found by actual experience that we were having cashflow trouble. And we're getting involved more and more heavily in commercial systems and in restaurant systems, background systems and things like that. In the case of the restaurants, we just had trouble getting paid on any basis. In the case of the commercial situations, there was just the reality that you had to give 30, 60, 90 day payouts, which were standard. And we got almost the same types of requests that Tim's talking about from the owners of these companies: "Can't you do something in the home?" And the business reasons were very compelling for us to do so. Over the course of the years now, it's about a $2 million a year operation and has grown very nicely. So, we saw it as a very real growth opportunity and one that lent itself to a small, service oriented way of doing business.

MORRISON: You've also worked with home developers.

STEVENS: Developers are a different category of people. It's a different service that's being looked for if you're serving a contractor for spec homes. When you're serving people in their home, you're providing music and entertainment readily available to

(continued on page 56)



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Ask Dr. Wokka

by Dr. Wilhelm Wokka

Dear Dr. Wokka,

Here in Dresden, Germany, we prefer to test the intelligibility of our column speakers by simply giving them to Professor Herr Doctor Wolfgang Amadeus Bachshundt at Die Instituen Von Akustikk Lautshprekkeren Verboten En Kollektenderdaten Teknikke, in Cologne. Dr. Bachshundt then subjects them to many complicated tests made on the most expensive and elaborate instruments in Europe and assigns them an intelligibility number; the famous "Sprechterfactten" number. Every loudspeaker in Germany used in a government or public facility bears this stamp of quality, so that the public will know exactly how much information it can expect to receive. Of course, we have a highly technological society here and everyone calculates the correct "intelligenfachtten" for the event they are about to witness. There is no leaving the question "What's he saying?" to chance. Everyone knows beforehand exactly how many words will be understood and how many won't.

So, my question is this: When are you Americans going to wake up and develop a truly scholarly and scientific method of measuring intelligibility and give up all the silly things you are now "playing" with? Right now, your speakers are so big that no university in its right mind would ever even consider trying to measure their intelligibility. You must make them smaller so that they can be measured in the University; our column speakers are ideal for this. After all, what good is a public-address loudspeaker if it is too big to have its intelligibility measured?

Lastly, why must you Americans insist on continuing your practice of sending your musicians over here to take our youths' money? Himmel! You and the Japanese. First cameras and now the Waltz!

Helmut Hottentottermann
Dresden

My Dear Herrer Helmut,

Intelligibility! Ah, such a noble subject to address. But first a few hatchets to bury. The waltz was invented in 1568 by the American Indians, not the Germans or the Austrians as we have been told. So was smoking cigarettes, which the Europeans seem to think they invented too, along with this pansy inverted-forefinger-and-thumb style.

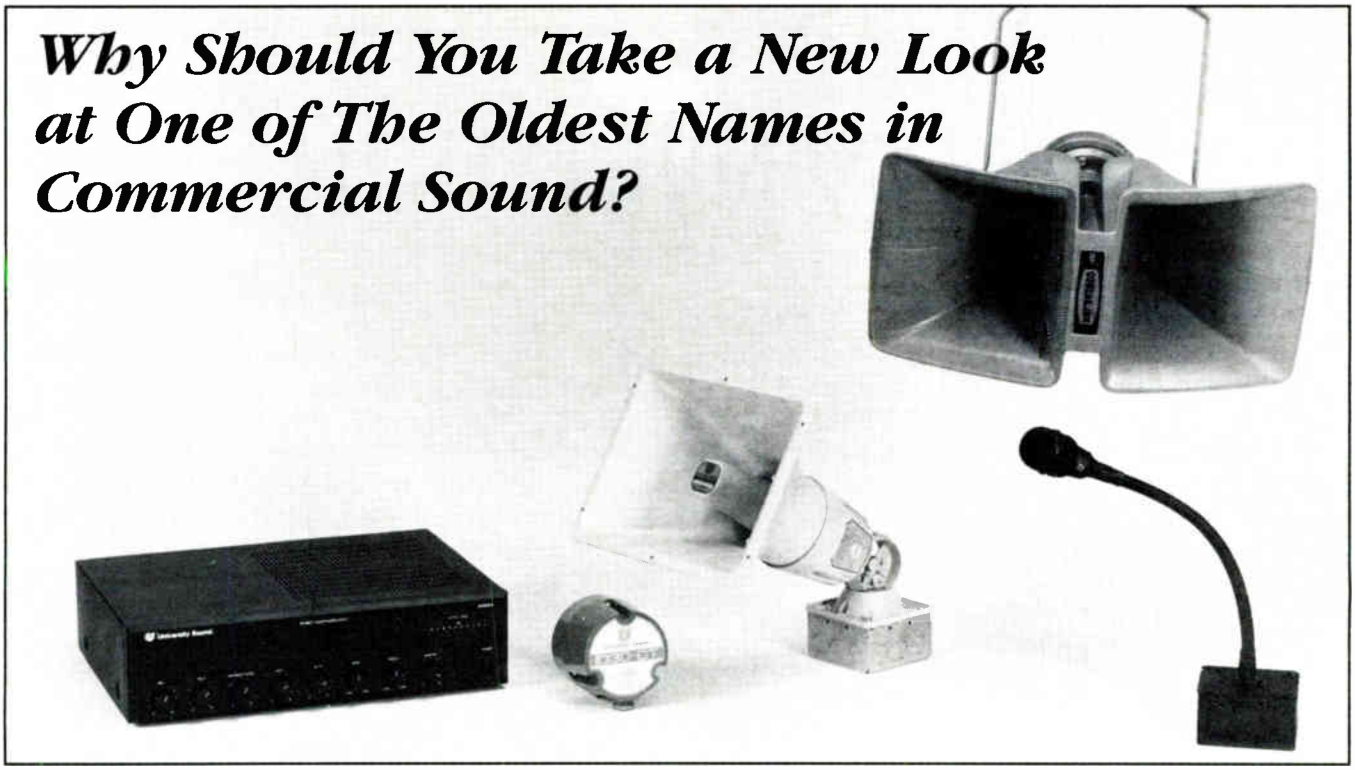
Enough culture clash. In America, we use the RASTA intelligibility number, described in this column many times. It involves not only the speaker and the room it's in, but the sound level and spectra thereof also. We use it mainly for purposes of confusing owners of large stadiums and wealthy sports teams into spending more money for things they really don't need but think they do in the resulting confusion. It's sort of a planned obsolescence, and necessary for the industry to continue to flourish. Of course, there are those who think they can merely hear the differences in intelligibility of speakers, but this is rubbish; scientifically proven falsehoods. Besides, this would only turn the business over to the masses and we would all be out of a job, n'est ce pas? Candid and radical talk here, but we're all in the same boat and we have to be honest about our lives' work.

Intelligibility numbers on speakers is a good idea in principle, but how about in America where the diet is so varied and the resulting speech is so different. Where else in the world can you have "Ha, y'awl," "Eh, baby, wass happ'n'n," "Hello," and "Yo, Vinny," all mean the same thing? Of course, in Europe, the French, Germans, Italians and English all sound different, a result of the various national diets per the famous work of my great grandfather, Wolfgang Hamburger Wokka XIV. Snails, bratwurst, garlic and limes just make people's mouths take on different shapes and attitudes when conversing. Thus, your intelligibility numbers will not work out of your country. What good is this? Don't you want to sell speakers outside of Germany?

Lastly, about your blatant and uppity slur on the size of American loudspeakers: We just plain have more room over here, and so do our universities.

Next month: A field trip to Testa Communications, magical kingdom of audiophonical journalizologists in ivorytower residence and so much more. ■

Why Should You Take a New Look at One of The Oldest Names in Commercial Sound?



Founded in 1936, University Sound holds over 100 patents and builds a broad line of commercial sound and interconnect products. As a leader in this industry, we are constantly improving existing products, and working hard to better serve our customers in every way.

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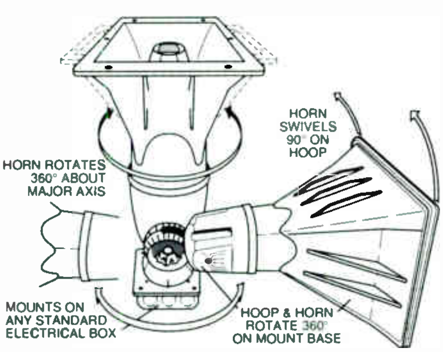

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University has also broken new ground with the US690 cardioid dynamic *gooseneck* microphone, the first commercial microphone using the N/DYM™ magnetic structure. Its super-strong Neodymium delivers 6 dB more sensitivity and permits a capsule geometry with lower peak distortion than the best of ordinary dynamic mics.

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THE AES CONVENTION

BY ALLAN VARELA

The Eighty-Seventh AES show, being held in New York from October 18th to the 21st, is shaping up as an important indication of the growth in technology for audio recording, audio for video, and audio for live performances. A large contingent from the sound and communications industry is expected to be on hand at AES's venues at the Hilton and Sheraton Hotels in mid-town Manhattan.

According to Don Plunkett, the executive director for AES, "The theme for this show is 'Audio for the Next Decade and Beyond'. The theme is fulfilled through the papers and the workshops. Also, there will be quite a few exhibitors with new products out."

Attendance is expected to be between twelve and fourteen thousand people, and, at press time, the exhibitors list exceeded 210 and counting.

"This is the largest show we've had for New York," explained Plunkett. "Because of space limitations, we couldn't fit into the Hilton. So now we're in the Hilton and the Sheraton." Exhibits will be at both hotels, with the Mullen Museum (history of audio) at the Sheraton. All the papers will be at the Hilton.

The AES, of course, is a technical convention, and a great deal of the attention at this show is devoted to the technical papers presented. Over seventy-eight papers are being presented this year on topics ranging from analog and digital audio electronics, to a new process for pulling ultra-pure copper wire. "We're very gratified that there are so many technology papers this year." says Ted Uzzle, the conference chairman. "That is to say, really new research; particularly papers from university researchers and also by consultants. We seem to have fewer papers from the new product category than in past years [papers that simply describe a company's product]. In fact we only have about five or six of those in the entire convention which is really quite exceptional."

Some of the sessions include: Analog and Digital Audio Electronics (two sessions)—Dennis Bohn, chair; Architectural Acoustics—Cyril Harris, chair; Audio and Acoustic Tests and Measurements—G.R. Thurmond, chair; Broadcast Sound—Bob Dixon, chair; Large Array Systems—Neil A. Shaw, chair; Perception of Speech and Quality of Music—Ann K. Nabelek, chair; Recording—Chips Davis, chair; Sound Reinforcement—Clay Powers, chair; Transducers—Mark Gander, chair; and New Directions in Audio—Tim Smith, chair.

The interest for the sound and communications community seems to lie with 3-D sound. "Certainly the hot topic with a great deal of research going on all over is the subject of three dimensional sound fields created by loud speakers," continues Uzzle. "We have papers from all over the world that deal with various aspects of that subject, so much so that we have created a new session category, to be chaired by Ron Streicher, to encompass them."

This new category is tentatively named New Directions in Multi-Channel Sound Reproduction (subtitle: If God had meant

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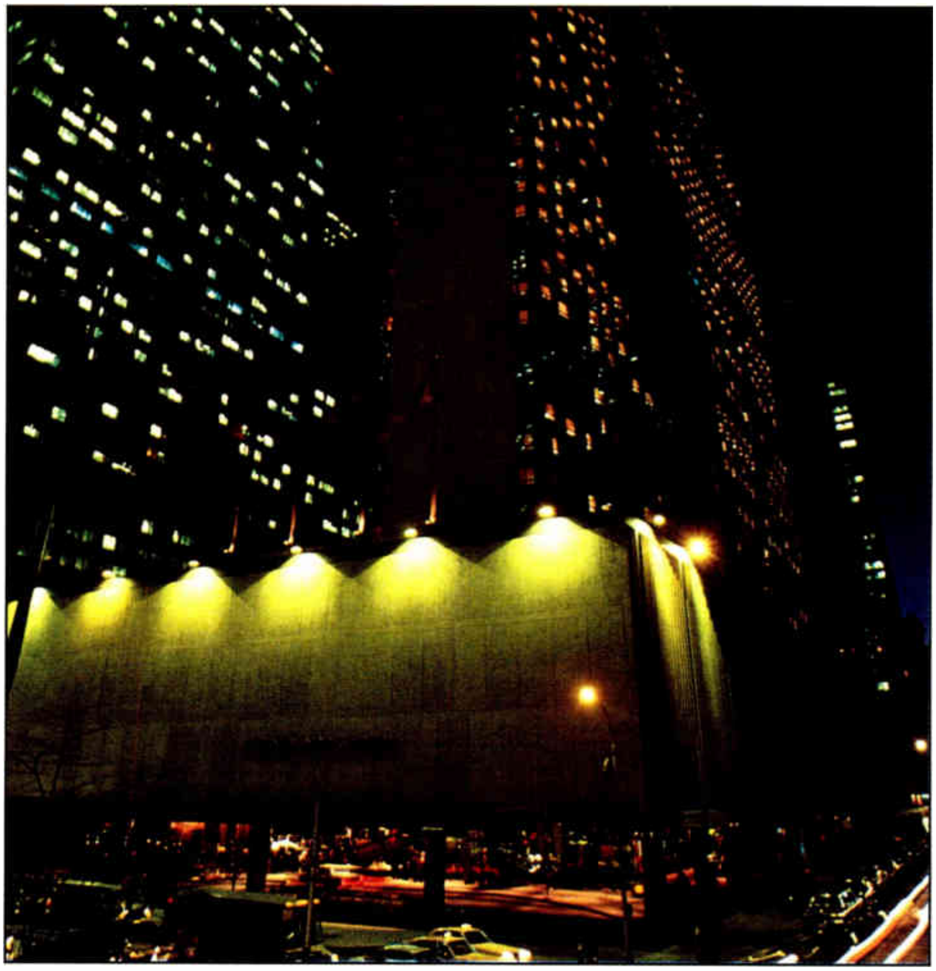

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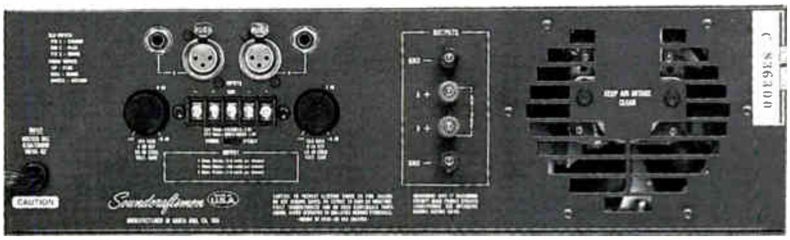

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The AES 87th Convention Schedule

Wednesday

MORNING

Business Meeting
Session A
New Directions in Audio
Workshop 1
Capturing the Live Sound
Workshop 2
Fiber Optics: The New Medium for Audio

AFTERNOON

Session B
Broadcast Sound
Workshop 3
Optimizing the Listening Environment — A Tribute to Charles Bilello — Noise Control, Room Acoustics, Spatial Hearing, Objective Measurements
Workshop 4
Video Sweetening Basics for Audio Engineers

EVENING

Session C
Sound Reinforcement
Workshop 5
Design for Studios & Facilities
Workshop 3 (continued)
Subjective Evaluation of the Listening Room

Thursday

MORNING

Session D
Speech Perception & Quality of Music
Workshop 6
Loudspeaker Design
Workshop 3 (continued)
Recording Studios
Technical Tour #1
Museum of the Moving Image, Astoria, Queens
Special Event #1
Cooper Hewitt Museum

AFTERNOON

Session E
Acoustics of Loudspeaker Arrays
Session F
Recording
Workshop 7
Preserving our Audio Heritage
Workshop 3 (continued)
Automobiles & Movie Theaters

EVENING

Session G
New Direction in Multi Channel Sound Reinforcement
Workshop 8
Teleconferencing & Board Room Audio
Open Meetings of the Technical Committees

Friday

MORNING

Session H
Transducers
Workshop 9
Education in Audio
Workshop 10 (10 am-5 pm)
Theater Sound (held at the Minskoff Theater)
Workshop 3 (continued)
The Record-Playback Cycle
Technical Tour #2
Radio City Music Hall
Special Event #2
New Botanical Gardens

AFTERNOON

Session I
Audio Electronics
Workshop 11
Ones & Zeros in the 90s
Workshop 12
Workshop 13 (continued)
Performing Arts Facilities & Arena

EVENING

Session J
Audio Electronics
Workshop 12
Sound in Difficult Environments

Saturday

MORNING

Session K
Audio & Acoustic Tests & Measurements
Workshop 13
Bruce Swedien: A Lifetime in th Recording Arts & Sciences — NARAS Educational Workshop
Technical Tour #3
Rogers & Hammerstein Archives, Lincoln Center

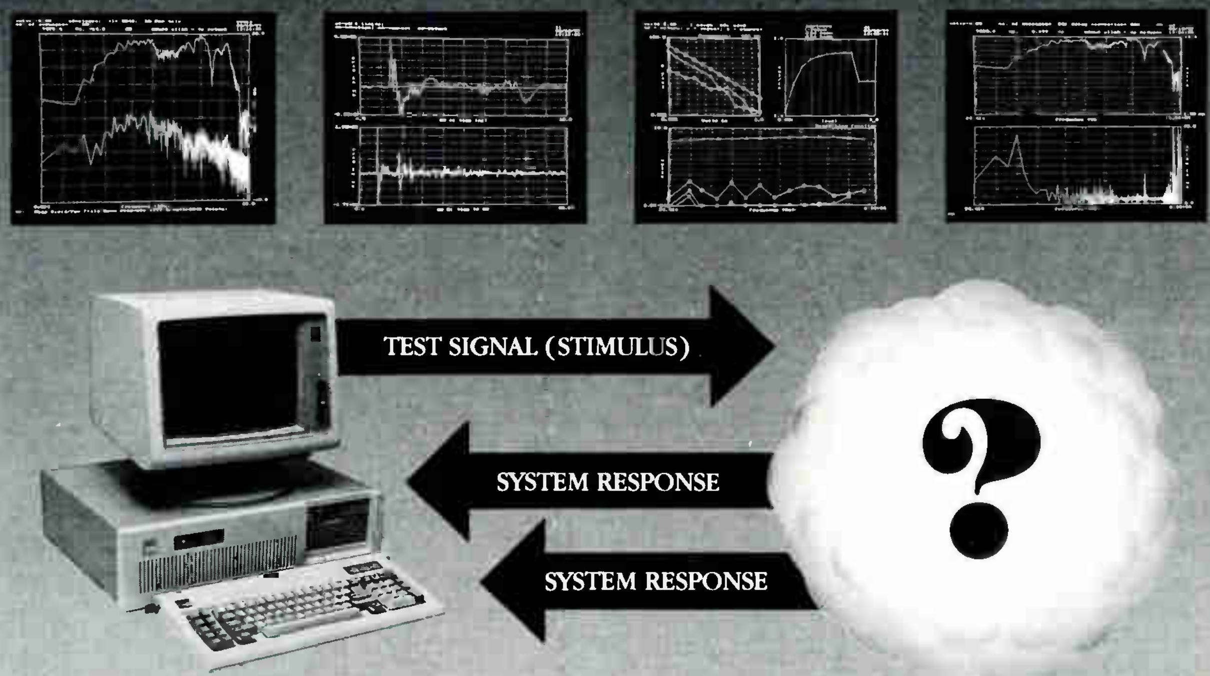
AFTERNOON

Session L
Architectural Acoustics
Workshop 13 (continued)
Bruce Swedien: Part 2

EVENING

Reception
Awards Banquet

Turn Your PC Into A Complete Acoustics Lab



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(continued from page 18)
us to listen to surround sound...), and its chairperson is enthusiastic about the future implications for the audio industry at large.

"The whole concept of multi-channel sound [Streicher's term for 3-D sound] is something that is growing in a number of areas of the audio industry," says Streicher. "This is something that is growing in stature in the film element, and, because of the tie in with motion pictures, the development of video tape and disc systems, and multi-channel sound playback systems for the home, it has now spread into the consumer market as well. Companies like Shure Brothers, Lexicon, and Yamaha are starting to come up with home 'surround' type systems. They are, of necessity, compatible with the Dolby system in one of their modes. But of course all of these systems have other modes of operation whereby, in one form or another, they synthesize some form of surround sound from stereo, and they even synthesize some form of stereo from mono. How they go about it is quite proprietary and quite diverse in their philosophies and applications."

3-D sound concepts have been incorporated since the forties with Fanta- Sound in Fantasia, OmniMax, I-Max, six track Dolby Surround, and the Holman THX system, among others. 3-D sound has found its way onto sound effect libraries, and is used on pop music recordings. Traditional stereo imaging involves left/right, and front/back perception. 3-D technology tries to create a total sound environment that surpasses the limitations of stereo imaging by adding up/down orientation, along with the total immersion of the listener in the audio field. The excitement about 3-D sound is that it effects everything up and down the line from studio design and new electronic processes, to home speaker design and listening environments; from more sophisticated movie theater environments, to 3-D applications in live theatre and music concerts. The intrinsic fecundity of 3-D sound for the audio industry is significant, and the

(continued on page 60)


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SPORTS STADIUMS What's Happening?

There's a renaissance in stadium construction, and for the next few years, business will be busy with new sports venues all across North America. What's happened recently in Stadium Sound, what are the challenges faced by the sound industry?

BY BILL THRELKELD
JBL Professional

Stadium Sound is an active part of the business, with facilities such as the Louisiana Superdome undergoing transformations.

One of the greatest challenges sound contractors face is designing and installing sound systems for a stadium, large arena, or outdoor concert facility. Great risks can be involved with such systems, due to the immense size and costs. But with the tremendous challenges can also come tremendous rewards.

"The challenges in designing any type of (cluster or individual speaker) distributed sound system are numerous," says John Monitto of Pro Media in El Sobrante, California, designers of the new system in San Francisco's Candlestick Park. "Should the system be direct-coupled or use a 70.7-volt distribution scheme? What type, how many and what time requirements will the audio delay units have? How much line loss can be expected in the long cable runs that are required? What type of control for low-frequency buildup or other problems will be necessary?"

Among one of the first challenges that must be addressed is where to install the speakers themselves. In a new facility, this can be more complex than in an older venue, because in an existing system there is a history of what has and what has not worked. There are of course two basic types of system designs for a large system of this type: distributed or central-cluster. The speakers may be "scattered" (at precise locations, of course) around the facility in a distributed system. In other cases, a cluster system can function as a central point source, usually at a scoreboard or end-zone location.

There are variations on these two types of systems, as Steve Romeo, Market Manager for JBL Professional, points out. "Occasionally, a client will opt for a true central system, suspended over the center of the playing field. Such a system was installed at McMahon Stadium in Calgary, Canada, for the 1988 Winter Olympic Games."

"The most challenging aspect of designing a stadium system is to create a theoretical point source from which all loudspeakers emanate," says Drew Serb, President of Pro Media. "The array must follow the point in both vertical and horizontal planes. All new arenas have some sort of central scoreboard, video screen, etc. Fitting properly arrayed loudspeaker clusters into these scoreboards is one of the biggest challenges faced in designing stadium systems."

Howard Smith, of Smith Fause and Associates agrees with Serb. Smith, Fause and Associates designed the new sound system recently installed at Denver Mile High Stadium. "The biggest challenge in designing a sound system for a large outdoor stadium is uniformity of coverage, particularly at low frequencies. On the Mile High Stadium job, the obvious choices were between an upgraded and modernized scoreboard cluster system and a high-quality distributed one."

For Mile High Stadium, Smith, Fause and Associates chose a distributed system, replacing an older scoreboard cluster. In a unique twist, once the speakers were removed from the scoreboard system in this installation, stadium management found that the resulting space could be marketed out to advertisers, and the extra advertising revenue generated from this use of space helped to offset the cost of the more expensive distributed cluster. And crowds at Mile High were noticeably impressed with the new system.

Jim Ford of Ford Audio talks more about some of the problems of large stadium systems. Ford Audio designed the system currently being installed in the Houston Astrodome. "The only problem (for the Astrodome installation) is that the reverb time in the low frequencies is between 15 and 20 seconds and once you start putting low frequency energy into the space, you run into problems.''

As mentioned earlier, the costs in designing and installing these large systems can be enormous. Owners will often sacrifice some sound quality in order to keep costs of the installation to a minimum. In most cases, a central cluster is the less costly of the two types. ''In stadium sound system design, a central cluster is always first choice,'' says Drew Serb. ''However, there are few stadiums that lend themselves to a point scource. Most stadiums do not have a location to place a center cluster without overpowering the audience closest to the loudspeakers. A high percentage of stadiums requires distributed systems. These systems are a major challenge to design due to problems with sight lines, camera angles, etc.''

Acoustical problems can be resolved in ways other than speaker placement. Such was the case with the new installation at Dallas Stadium, home of the Dallas Cowboys. K.R. Dickensheets and Associates, of Austin, Texas, was contracted by the city of Dallas to study the job. Dallas Stadium is an enclosed facility, which can create more nightmares through reverberation and delay problems. ''Like most large rooms, reverberation, focused echos, discrete echos and noise are a problem,'' says Dickensheets. ''An attempt to control these problems was made when the stadium was being built by applying spray-on acoustical cellulose to the entire underside of the roof structure ... the reverberation problems are now virtually gone.''

And what role does the contractor play in these stadium jobs? Drew Serb answers. ''Recently, the contractor appears to be playing a much larger role due to time and dollar restraints. The consultants are being asked to provide performance specifications, therefore requiring the contractor to provide the detailed design work.''

Howard Smith sums up his feelings on designing large systems. ''The key to a successful design lies in understanding the limits of different types of systems, and the tradeoffs involved. You must be willing to find out how each type of system will work in a particular stadium's geometry.''

GOOD SOUND MAKES THE RACES EXCITING AT RIVER DOWNS

BY PAUL CLARK

At Cincinnati's recently rebuilt River Downs Race Track, the sound system carries the track announcer's call throughout the 178,700-square-foot grandstand building. The customers hear the announcer through 195 speakers in the grandstand, apron, paddock and pavillion, the two mezzanines, the pressbox and the upper and lower clubhouses.

The primary system carries the track announcer's voice throughout the public areas. There are four local sub- systems, each of which can carry either the primary feed — or exclude it in favor of a local feed that's remotely switched. The primary system also carries a television audio feed from a satellite. The video shows up on 275 monitors, bringing out-of-town races into River Downs and replaying the local races.

''The way we supply information to the public has been greatly improved,'' wrote Jack Hanessian, general manager, in the souvenir program booklet introducing the new River Downs. ''The sound system is all brand new, with speakers spaced to improve on the hot and cold spots that were a problem in the past.''

The point is this: audio commuhication is marketable. It helps the client attract customers; it helps the customers enjoy the races. Clear sound encourages the customers to come again and again.

THE MODULAR SOLUTION

Even in a 178,700-square-foot building, equipment-rack space is at a premium. So we were attracted by the compactness of Industrial Research Products' System 41, in which a 10-inch-high rack-mounted mainframe, DJ-4101, contains a monitor and space for 13 signal-processing modules. Our System contains only six.

So it is easy to expand. It is easy to secure (behind a lockable panel) and easy to service; also easy to adjust, since all the controls of all five mixers are within a few inches of each other. The available rack space here is minimal; if we hadn't selected a modular system, we would have needed one more rack. Our five System 41 DJ-4102 mic mixers all have remote-switching capability. We use it on four of the mixers.

The fifth handling the primary feed has two mic inputs at the track announcer's position. Input 3 is from a wireless mic system, allowing the announcer to call the races from the grandstand; input 4, from a mic at the winners circle. Input 5 is switchable between an audiocassette player and an audio cartridge machine that plays the traditional bugle calls; and input 6 is the audio feed from the River Downs television studio.

Of the mixer's two outputs, one goes to the video studio, from there to the monitors and replay recorders. The other goes through System 41's DJ-4110 limiter module. If the announcer gets excited and raises his voice as the race progresses and the horses thunder down the stretch, the limiter module puts a ceiling on the gain and helps avoid overdrive and distortion.

SIGNAL PATH

The limiter's output moves through four Symetrix 571 SPL computers, ambient-sensing automatic level controllers that change the gain in proportion to environmental noise conditions. Sensing mics are in the listening areas. When the crowd roars, the gain increases.

The signal moves through the level controllers and equalizers to the grandstand speakers (two-thirds of which are on time delay), to the apron horns, and to speakers in the mezzanines, clubhouses, pavillion and pressbox. The third floor east mezzanine, third floor west mezzanine, upper clubhouse and lower clubhouse are the areas that have the local sound systems, each with its own System 41 DJ-4102 mixer module. In each of the areas, a remote switch can shut out the primary feed (which comes in through input 6 of each mixer) and activate a local mic input.

The upper clubhouse sub-system, for

(continued on page 60)

The Louisiana Superdome Sound System Upgrade

BY STEVEN M. HODGE

When the Louisiana Superdome opened 15 years ago, the sound system was state of the art. Jacek Figwer, then of Bolt Beranek and Newman, designed the original system to handle a range of events including football, baseball, basketball, boxing, conferences, and conventions. To accommodate this range of events, he provided four configurations, each based on the sound originating from a given location. Relays assigned the delay channels to the various speaker zones. Switches on 125 speaker lines provided additional control. This system worked well providing the event was centered on one of the locations included in the original design and the operators turn on only the appropriate speakers.

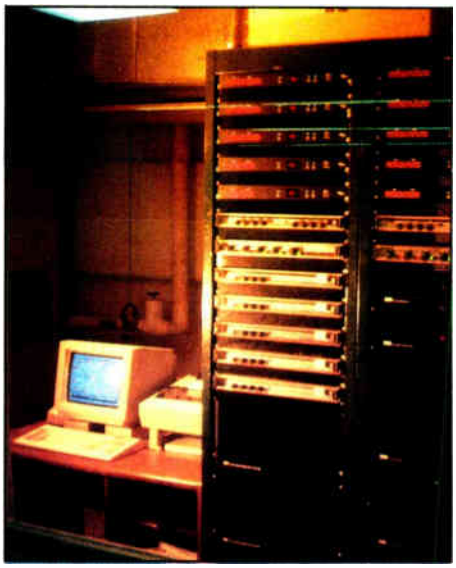
Over the years, a number of events took place with a stage in a location the sound system was not designed to support. This meant that either a sound system had to be brought in that would cover all the seats or, all too often, the existing system was used in conjunction with a portable system, even though the delay patterns would be wrong, causing unintelligible sound in parts of the Superdome.

Late in 1986, the management of the Superdome felt it was time to start addressing some of the problems they were having with the sound system, including reliability. The delays, equalizers, and line amplifiers in the system were failing fairly regularly and, because of the size of the room and the number of speakers, the failures would go unnoticed by the system operators. Complaints about the sound were becoming more common.

The S. M. Hodge Co. was hired to evaluate the existing system. After the system was repaired to bring it up to its original performance, planning was started to upgrade the system based on the following goals:

- Make the system more reliable and provide a reasonable way to determine if the system is functioning.
- Make it easier for system operators to set system configurations.
- Provide for a wider range of configurations.
- Keep the cost of the upgrade as low as possible, always a consideration.

To keep cost down, it was decided to reuse the existing speakers and amplifiers. The original design provided reasonable coverage and, as the adage goes, "if it works, don't fix it." The distributed



speakers were rewired to provide 24 zones instead of the original 10, requiring the addition of approximately five miles of new speaker line. Each speaker line leaving the ammplifier room (250 lines total) was routed through a DPDT relay to provide for speaker switching and to provide access to the speaker lines for testing.

The existing delay units were replaced with new units providing a total of 26 channels of delay. All the original equalizers were replaced. Because the power amplifiers in the system operate with very little headroom, front end processing was added. This included a compressor in each audio path and an enhancer and multi-band limiter in the main systems.

AN IED computer-controlled matrix was selected to replace the line amps and relays used to assign the delay channels to the various amplifiers. This system provided the most flexibility because the different configurations are entered into the computer rather than wired in or represented by pins in matrix boards. The IED computer that controls the matrix also turns the speakers on and off using the relays in each speaker line.

The consultant enters the system configurations, giving each a name. A total of 50 different configurations can be stored. The system operators select a configuration from a menu of the names, and the computer assigns the delay channels and turns on the required speakers. The operator has the ability to override the speaker on/off function but not the assignment of delay channels. The computer shows a graphical representation of all the speakers in the system, with the color of each group of speakers showing the assigned delay channel. Speakers not turned on appear as an outline only.

The IED equipment also provided the solution to the problem of determining if the system is functioning properly. This is done in two ways. The output of every active component in the system (160 points) is hooked to a computer-controlled monitor selector. From the computer keyboard, any of these points can be accessed with the audio signal played through a monitor speaker and the audio level displayed on the computer screen. The listening level is preset for each monitor point in the system so that no matter what level the source is, the listening level is reasonable.

The IED equipment also can carry out its own check of the system by putting a tone into the system, reading each monitor point, comparing the level at that point with a stored value, and report any deficiencies. In addition to checking all of the active components, the Superdome system is unique in its ability to also check all of the speakers. Using the speaker line relays mentioned earlier, it is possible to

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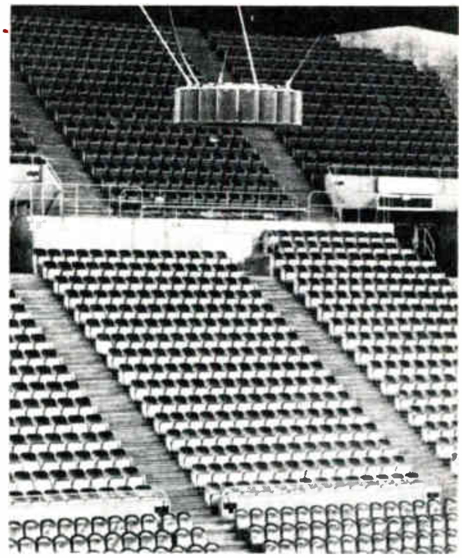
OAKLAND ARENA INCREASES INTELLIGIBILITY

A new system in the Oakland Arena, home of the Golden State Warriors, was designed to increase the gain and intelligibility while adding special audience participation audio effects. San Francisco-based Black Boxes, Inc. installed more than 32 Apogee loudspeakers and attendant processors. Since the installation took place just prior to the NBA playoffs, Apogee and Black Boxes, Inc. worked in the early morning hours a week before the playoffs to install the sound system without interrupting regular season play. The reason for the upgrade was to replace a system that was inadequate for the increased attendance at Oakland Arena. The business aspects of sports arenas impact upon the sound system. The Golden Gate Warriors' Director of Events and Game Operations, Craig Van Schotenstein, commented, "It was extremely important to the ball club that fans at all levels of the

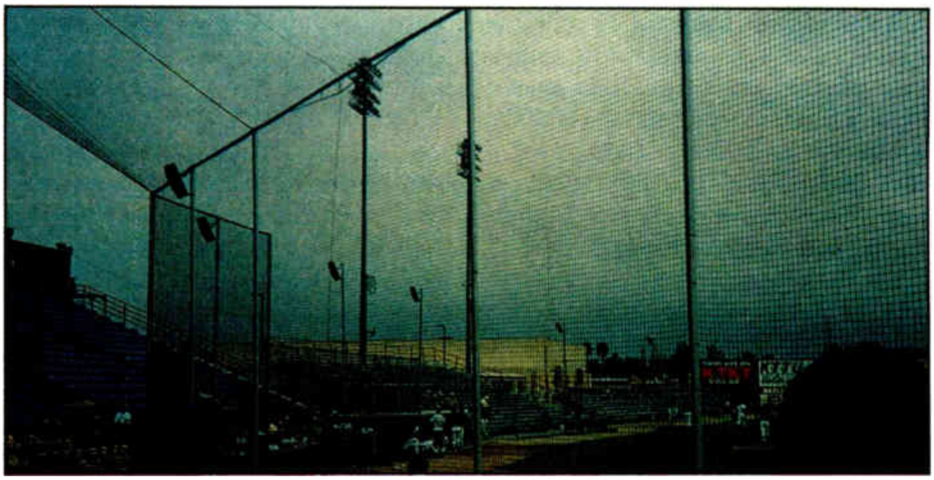


arena clearly hear the message of our sponsors." To achieve the "desired gain and intelligibility in the stadium," 18 AE-5 Apogee loudspeakers with attendant processors were mounted on a circular ring under the scoreboard in the center of the arena. Sixteen AE-2 wide angle, low profile speakers were installed around the stadium 75 feet out from the center cluster in a concentric circle as a delay system which provided additional gain to the upper levels. Six AE-12 subwoofers were mounted above the scoreboard, providing bass response for special audio effects. The system also featured Apogee amplifiers, a Klark Teknik parametric equalizer and digital delay line, and a Yamaha mixer. Phil Bailey, Black Box Inc. engineer commented that the speakers were easy to install by contractors and technician sat the work site. And the Oakland Arena Assistant General Manager, Bill DeCarlo, said, "I only wish we had installed the system sooner."

TORO STADIUM GETS SYSTEM

BY WES ALDERSON

The Tucson Toro Stadium is used for the minor league Tucson Toros and also as the spring training location for the Cleveland Indians. Chambers Electronics in Phoenix installed a new sound system at the stadium. Originally, the clients wanted a central cluster horn configuration. Chambers used a Modeler design, along with their own knowledge of sound, to predict that a Bose 402 system, mounted on light poles and foul screens, could be used to cover the crowd as well as possible. The 402s deployed in this manner around the crowd also provided a minimum of sound pressure level behind the seating area in the surrounding residential neighborhood. This aspect of coverage, predicted by Modeler, was a distinct advantage compared to a central cluster, since one generally does not want to blast the local neighbors with the ballpark PA system. This is a 70 volt system, which re-



quired the use of an internal wealther-proof 70 volt transformer at the location of each 402, due of course to the necessity of 70 volt line. A direct coupled system running out over the perimeter of an entire ball park would have suffered excessive loss. The system is installed and operating. It performs exactly in the manner predicted by Modeler. In addition to high SPL in the seating areas, the 402 system provides clarity of sound a wide response curve. Wes Alderson is with WesTech Marketing in Phoenix.

TUCSON TOROS EQUIPMENT LIST

16 Bose 402 speaker; 16 Altec 15732 transformers; 16 Omni Mount mounting brackets; 1 Bose 402E controller; 2 Altec 1270 amplifiers; 1 Altec 1712A compressor limiter; 1 Altec 8558A programmable equalizer; 4 Altec 15470 output transformers; 1 Altec 1692-B 6-channel mixer; 1 Altec D648 dynamic mic; 1 Atlas DS-7 mic desk stand; 1 Gold contact spring loaded switch; 1 Telex FMR50 hand-held wireless mic; 1 Soundolier ACR-193 switchable power; 1 Yamaha C-300 professional cassette; 750 Choates ¼-inch air craft cable; 1 Soundolier 200-77 floor equipment rack; and 3M West Penn 225 16 gauge speaker wire.

YANKEE STADIUM AUDIO

Mark Hansen of PEM Electric in control location at Yankee Stadium.

According to Steven Schlaff of Norcon Electronics, when the sound system at Yankee Stadium was upgraded in the seventies, David Klepper's specs weren't completely adhered to. Since that time there have been several more upgrades — bringing the system back in line to the original specifications.

Pem Electric was the electrical contractor and Norcon Electronics the sound contractor at Yankee Stadium, which has just completed a further upgrade, done quickly and efficiently to the demands of Yankee Stadium, with manufacturer personnel such as John Wiggins of Community and manufacturers rep Vinny Macree personally helping out on site with demos.

Eight low frequency cabinets were replaced with Altec 817B weatherized black LF horns. These horns are manufactured of plywood and finished with a black lacquer followed with a urethane weatherproof coating. Two layers of screening were added to the front to protect the unit from driving rain. Each of the low frequency cabinets contain two 70v matching transformers and two 515-8G-HP 15-inch drivers.

Two equalizers and crossover network were replaced with new items. The new equalizers are Altec 8551B programmable, and the new crossover is a Urei 325. Additionally, the press and luxury box sound systems were replaced with new Altec 1715 and 2271 units. To better control the balance between the outfield sound system and the main speaker cluster, an additional compressor limiter was added to the system.

An "amplifier fault" panel was installed in the console room. The panel is wired to give a remote indication if any of the incremental amplifier modules fail.

Four Community M4/PC1564M/PCMX mid frequency horns with matching transformers were installed. Additional 1180 and 2271 amplifiers are required. Four horns, two amplifier frames, 16 amplifier modules, four matching transformers and one crossover are incorporated into the system.

The Yankee Stadium system also includes a CD player, a phonograph and a Muzak system.

THE YANKEE STADIUM AUDIO SYSTEM

Control Room Electronics

1 Yamaha M512 Mixer; 1 Tascam 122 cassette deck; 2 ITC-3M Omega Cartridge Player; 1 ProTech DA1528 Audio Distribution Amp; 1 ProTech 663CL Limiter; 1 Altec 1712A Limiter; 1 Altec 1612A Limiter; 1 Bogen TP100 Tuner; 1 Telex 230 Reel to Reel Tape Recorder; 1 Telex RP85 Preamp; 1 Kyocera CD Player; 1 Technics SLQX300 Phonograph; 1 Muzak BGM Player.

Mezzanine & Corridors

5 Altec 2280A Incremental Frame; 36 Altec 2271 Incremental Amplifier; 3 Klark Teknik DN700B Time Delay; 4 Klark Teknik DN701B Time Delay; 1 Altec 8051A Equalizer; 1 Altec 1606B Press Amplifier; 1 Norcon Amp Fault Panel; 1 Altec 8551A Equalizer; 1 Altec 1715A Press Amplifier.

Outfield Electronics & Amplifiers

7 Altec 2280A Incremental Frame; 52 Altec 2271 Incremental Amplifier; 1 UREI 525 Crossover; 1 Altec 8051A Equalizer Field Page Amp.

SPEECH INTELLIGIBILITY: PREDICTIVE TECHNIQUES AND CAD

BY MIKE KLASCO

What follows is an examination of intelligibility prediction techniques, a capsule survey of sound system engineering software that uses these predictive measures, as well as a brief look at the intelligibility testing instrumentation currently available.

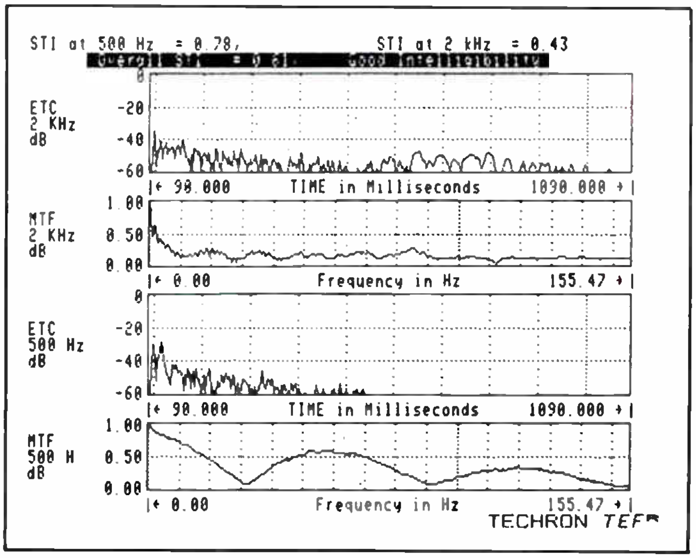
The main reason for installing a speech reinforcement system is to make it possible for the audience to better understand the talker. Speech communication implies not just intelligibility, but also comprehension and even retention of information. The integrity of the communication channel can be corrupted by distortion, as the spurious artifacts of the sound system such as hum, buzzes, background noise, ringing, etc. will degrade the quality, possibly increasing listening fatigue and thereby reducing the attention span of the audience. Listening fatigue may be contributed to by the increased mental processing effort required for comprehension as the listener's brain sifts through the distorted message to extract the signal. These factors can have a subtle effect on how the message is perceived by the listener, including reduced comprehension, attention, retention, motivation and credibility.

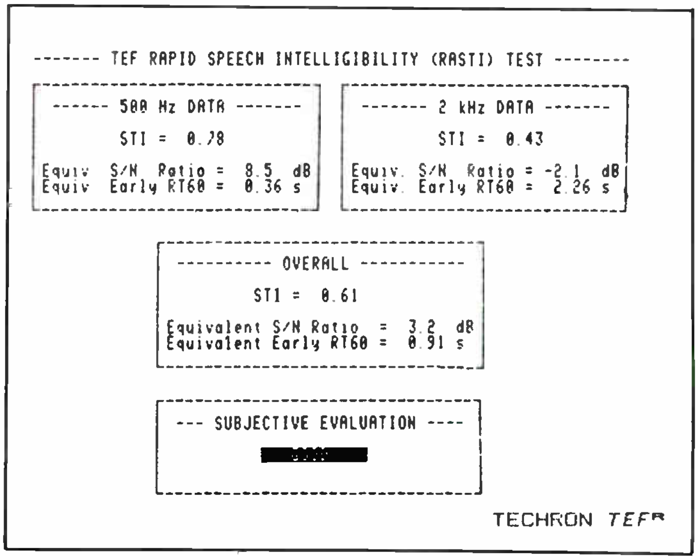
Speech understanding is a complex process that may be modeled as a hierarchy of layers. The bottom layer is the acoustic structure, phonemes (the sounds of speech); moving up to the next layer is surface features, then syntax, deep structure, and semantics; and finally upward to concepts. Speech understanding can be bottom up, top down, or any other combination. For example, in a room with poor acoustics, even through not every word is understood, the listener is able to decipher a lengthy message through the overall content, but at the cost of higher listening fatigue and shorter attention span.

It has been suggested that the clarity of the communication channel can be measured by the processing time required for a cognitive response of a listener. That is, the speed with which a listener could correctly respond to a question would indicate the efficiency of the signal transmission. This procedure would accurately weigh such factors as consonant loss versus horn throat distortion (as with a direct radiator speaker compared to a horn). Unfortunately, consultants and sound contractors rarely have the luxury of conducting time intensive testing programs to evaluate the effectiveness of the sound reinforcement systems they design. Rather, they must rely on some quick measure of speech intelligibility, instead of attempting to take a real measure of comprehension, listening fatigue, and so on.

PREDICTION OF SPEECH INTELLIGIBILITY

Speech intelligibility techniques include Articulation Index (AI) and Alcons, and are based on measurements such as background noise, reverberation, direct and reverberant sound level. Another technique is based on the modulation transfer function; Speech Transmission Index (STI) and a derivation, Rapid Speech





Transmission Index (RASTI) measures intelligibility more directly.

With STI and RASTI, effects of both background noise and reverberation are automatically included in one measurement. The measurement is made with both the test signal and the background noise present. While AI and Alcons can be predicted by formula, STI and RASTI do not lend themselves easily to predictive measures.

Each sound system engineering software developer has taken a somewhat different approach to intelligibility prediction. A few have omitted this function, others offer a few choices, and some provide enough raw data so the user can derive his own results, depending on what formula he is comfortable with.

JBL's CADP

CADP uses a qualitative rating system based on a modified Alcons formula. The reverberation time is compared with the direct-to-reverberant ratio in making an estimation of *excellent, good, fair,* or *questionable* intelligibility prediction.

The reverberation time is compared with the direct-to-reverberant ratio in the calculation of the rating. This estimate is based on the work of Peutz, and it is assumed that the system has adequate gain and that the signal-to-noise ratio is at least 25 dB. Peutz further assumes that the reverberant pattern is fairly normal across the frequency band and that there are no pronounced reflections in the space. In my experience, very few stadiums and arenas meet the 25 dB signal-to-noise criteria, and many smaller jobs do not have a true reverberant field and suffer from flutter echoes.

In any case, if you don't like these assumptions, CADP also provides you with direct/ reflected ratios for all seating locations, for both empty and full crowd conditions, so you can do intelligibility predictions your own way, if desired. When the intelligibility module is selected, all seating areas are calculated automatically, rather than one location. This a distinct advantage over other software that requires each location to be selected and displayed.

JBL's CADP II

JBL's new sound system design program will not debut until early next year, but the program's approach to intelligibility prediction will be to offer a full range of options to the operator, rather than to limit the choice to a specific technique.

UMBULUS

Although some earlier versions of Umbulus included an intelligibility module, the current release of the program does not attempt to predict this factor as the program developers feel there is too much disagreement on the accuracy of the available algorithms and the subjective nature of this measurement. The raw data are available within Umbulus to derive Alcons.

PHD

The PHD program offers the most comprehensive intelligibility module presently available. Alcons is provided, and also a conversion to RASTI. The user can specify the signal-to-noise ratio, and use different values for various locations. Results are shown for intelligibility using only direct sound as well as consideration of both direct and reverberant energy. The RASTI conversion uses a look-up table to convert from Alcons to RASTI. As RASTI is considered a fairly accurate measurement technique, but Alcons is considered by many as questionable, a simple conversion from an Alcons prediction to RASTI does not make the data any more reliable.

ACOUSTACADD

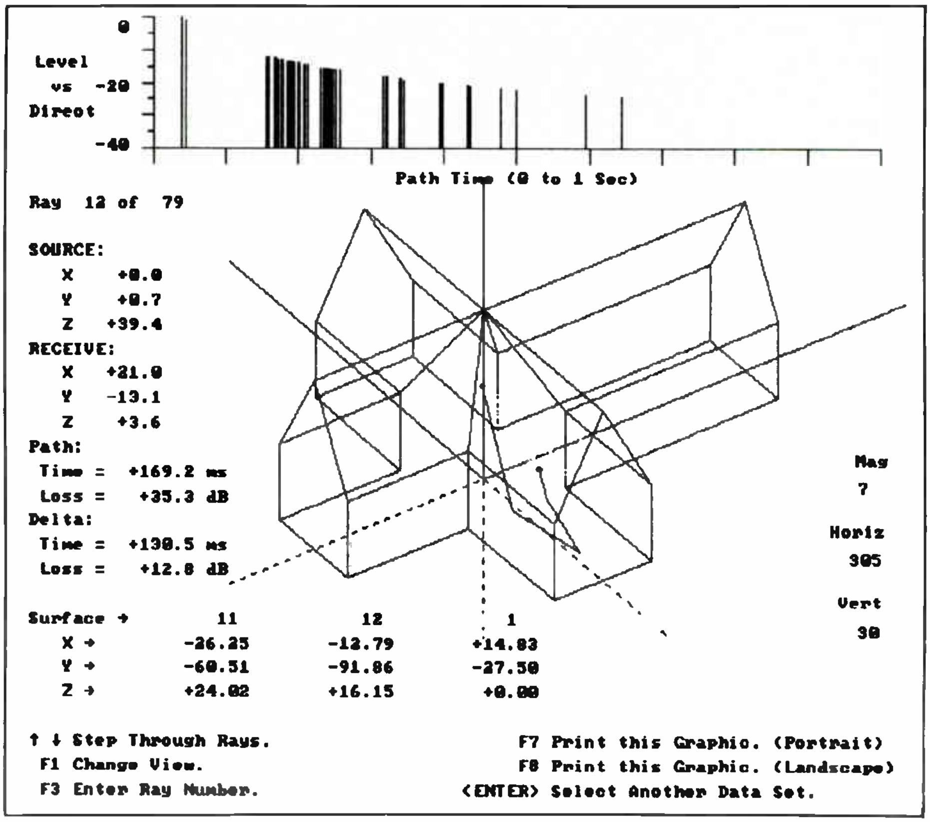
Release 1.0 of Acousta-CADD includes Alcons, but some sort of STI measure is contemplated for future releases.

A frequency band is selected and the user selects the on/off status of the speakers, and points at a location. The program calculates the Alcons for that data point. A summary of the Alcons for all the data points selected can be printed out.

Additionally, it is also possible to display an ETC type display showing the magnitude and arrival time of the energy from all loudspeakers with ON status reaching this point. Power and delay settings as well as off axis sensitivities are considered. Assuming the room model is created in enough detail, and the ray tracing algorithm is sufficient to derive a complete enough ETC curve, then the STI could be calculated. Another (more accurate) approach would be to provide a file transfer capability from an analyzer that could acquire the actual energy time curve from the room (such as a TEF, Ariel, or DRA MLSSA) instead of attempting to synthesize the ETC.

BOSE MODELER

Modeler presently does not attempt to predict intelligibility, as Alcons is not considered an accurate method by Bose engineers (among others). A new algorithm for predicting the speech intelligibility of sound systems in advance, which exploits the room and sound source models in a Modeler design file, has been developed and tested. A preliminary ex-



Mark IV Audio's Acousta-CADD. Typical screen of statistical ray tracing. Courtesy Altec Lansing.

periment by Bose engineers has shown that the new algorithm is more accurate than Alcons.

The new algorithm for predicting intelligibility is a derivative of another development that uses the room decay curve for a sound system in a room. However, predicting the energy decay curve requires tremendous model sophistication and computation power. The modification to this approach at Bose is called the Hybrid Energy Decay Curve (HEDC). This is a simpler and computationally more efficient representation of the energy decay curve consisting of direct sound, early reflection and late exponentially decaying sound.

HEDC contains three components: the direct sound, early reflections, and late exponentially decaying reverberation, all at some specified location.

The level of the decaying reverberation is extrapolated from the early reflections. The late part is spliced to the early reflections 210 ms after the arrival of the direct sound.

The energy in the HEDC is divided into two parts of the purposes of predicting intelligibility: early (or useful) and late (or detrimental) sound, using 80 ms as the dividing point. There has been a good deal of discussion on whether including the first 80 ms of early reflections as usable for intelligibility is a little too generous.

Although papers on intelligibility by Bose engineers have been appearing for almost five years now, none of these predictors have yet been introduced into Modeler. Part of the reason for the delayed addition of this function has been the controversy on the effect of speaker directivity and early reflections on speech intelligibility that has gone on between Don Davis and Bose engineers. Perhaps the simulation and testing program of this technique will enable Bose engineers to refine it to their satisfaction and intelligibility prediction will

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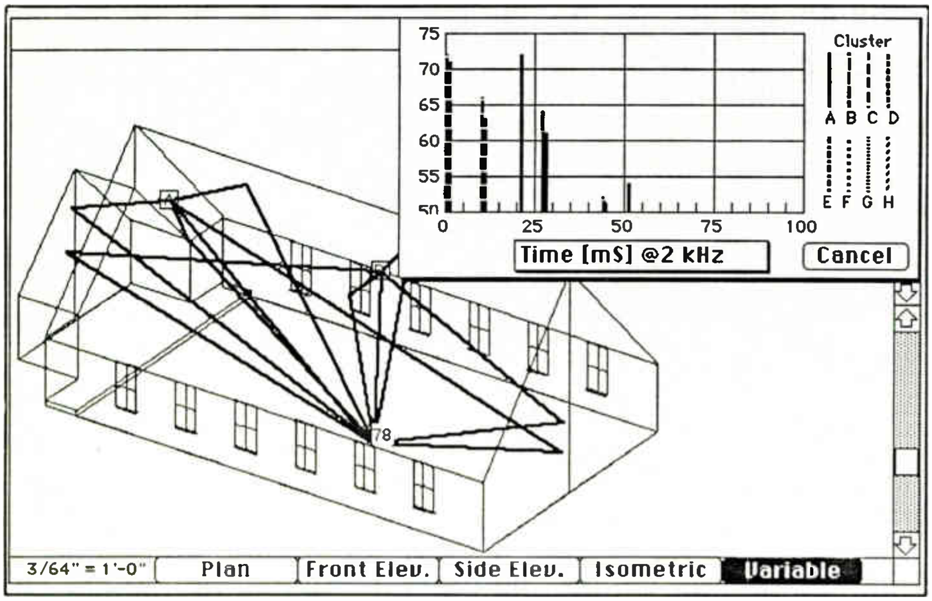


Bose Modeler. Courtesy Bose.

(continued from page 34)
be included in Modeler. Ken Jacob's paper in the December 1985 AES Journal ("Subjective and Predictive Measures of Speech Intelligibility — The Role Of Loudspeaker Directivity") and a conference paper from Birkle and Jacob given in the 1988 AES conference on Sound Reinforcement, "The Latest Advances In Predicting Sound System Performance In Real Space: Combining Intuitive User Interface With Acoustically Relevant Output," describe the Hybrid Energy Curve. Additional papers from Bose engineers are also referenced in these articles.

The Modeler runs on the Mac, and spectrum analyzers from Spectral Innovations and others will plug into the Mac II buss. If a file transfer capability was developed between the analyzer and Modeler, then the room decay curve from existing spaces could be imported into the program rather than attempting to synthesize it.

For another perspective on intelligibility, Don and Carolyn Davis' AES preprint from the November 1988 convention, "Application of Speech Intelligibility To Sound Reinforcement," provides an overview of their 1986 Workshop on Intelligibility and evaluation of the various intelligibility predictors and test equipment.

INTELLIGIBILITY UTILITIES

You do not need to buy a complete sound system engineering software program in order to get an intelligibility module. (Incidentally, the TPM software mentioned below is part of a set that also includes other programs for reverberation prediction, transmission loss through walls, and road noise.)

ALTEC L+I

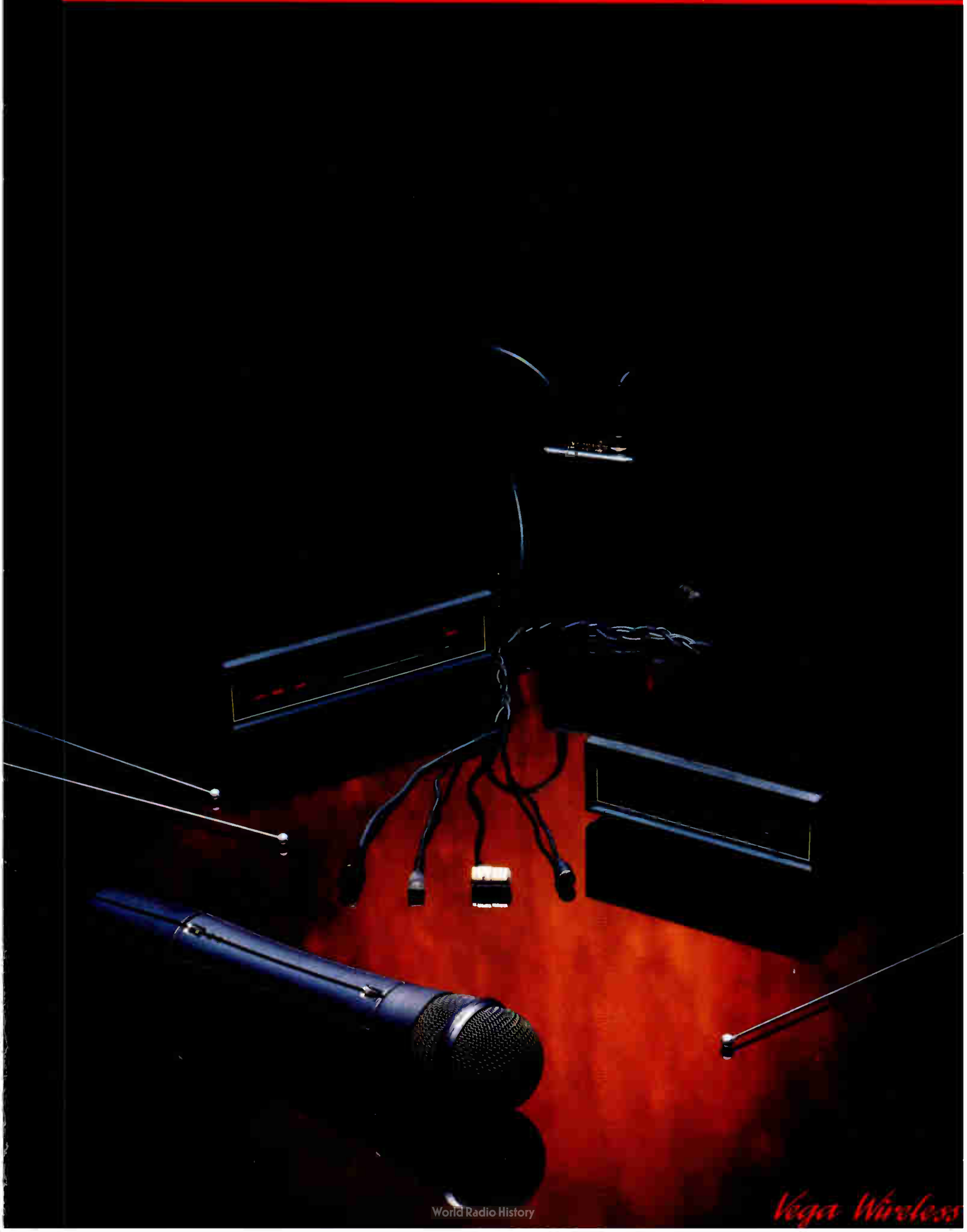
For those of you who still have HP 41 calculators, Altec's L+I utility was the first articulation loss program that I know of that was available to sound contractors.

TPM AISI

Articulation Index (AI) and Speech In-
(continued on page 41)

Ranger Wireless Systems

Vega



Ranger Wireless Systems

Ranger wireless microphone systems are Vega's answer to competitively priced wireless systems without compromising audio performance. They offer a higher signal-to-noise ratio and wider dynamic range by virtue of their exclusive CVX™ audio processing. You receive smooth, clear, virtually noise-free audio performance.

Ranger is available in both true-diversity and non-diversity configurations, and delivers exceptional RF performance through the transmitter's high radiated power and its sensitive, highly selective receiver.

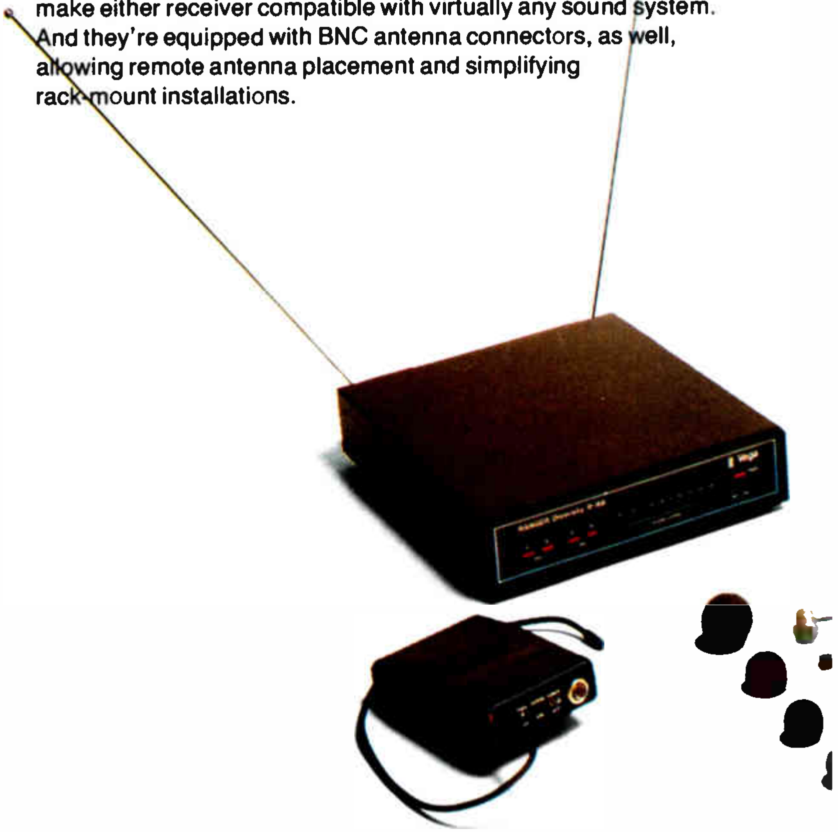
Ranger offers a wide variety of lavalier microphones and handheld microphone elements, operates on crystal-controlled frequencies in the 169-216 MHz range, and comes with a three-year warranty.

Ranger R-98 true-diversity/ R-97 non-diversity receivers

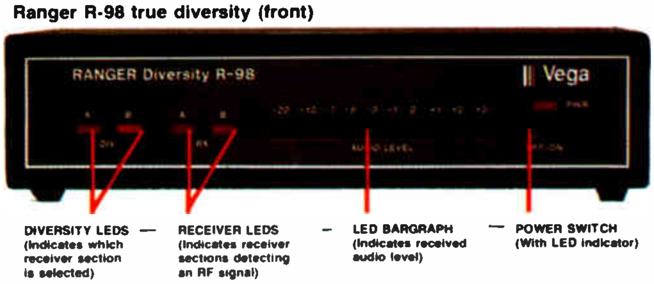
The R-98 and R-97 are virtually identical in features; however, the R-98 is a true-diversity design while the R-97 is a high-performance, non-diversity configuration. Both feature CVX audio processing, which provides lower distortion, a higher signal-to-noise ratio, and wider dynamic range than similar competitive systems.

Both receivers include sensitive MOSFET amplifiers and multipole LC RF filtering, for greater rejection of interfering signals. Extensive IF filtering is also provided, allowing you to operate 15 or more systems in one location.

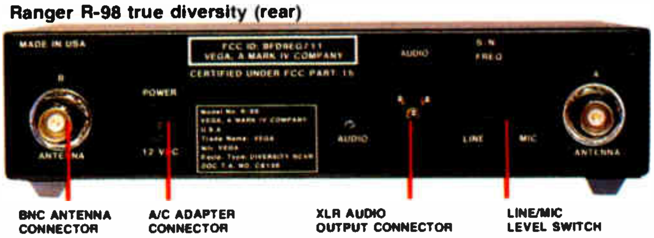
The R-98 and R-97 feature switchable, adjustable mic/line audio outputs via industry-standard XLR connectors, which make either receiver compatible with virtually any sound system. And they're equipped with BNC antenna connectors, as well, allowing remote antenna placement and simplifying rack-mount installations.



Ranger R-98 true diversity (front)



Ranger R-98 true diversity (rear)

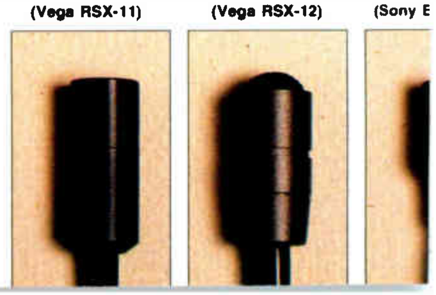


Ranger R-97 non-diversity

LM-206X — omni (Vega RSX-11)

LM-203X — uni (Vega RSX-12)

LM-201 (Sony E



Ranger T-93 bodypack transmitter

The T-93 bodypack is built for ruggedness and reliability. It's also equipped with CVX audio processing, providing you with clean, transparent sound.

The T-93's microphone level adjustment allows you to easily optimize your system's performance with different performers, as well as assuring complete compatibility with virtually any lavalier microphone. Lavaliers offered through Vega include the Shure 838, the Sony ECM-44B, the Crown GLM-100/E, and our own RSX-12 and RSX-11.

The T-93 delivers a full 50 mW of RF power, and its efficient antenna design ensures maximum operating range. Also, a silent on/off audio mute switch allows the mic to be turned on or off while the RF remains on.

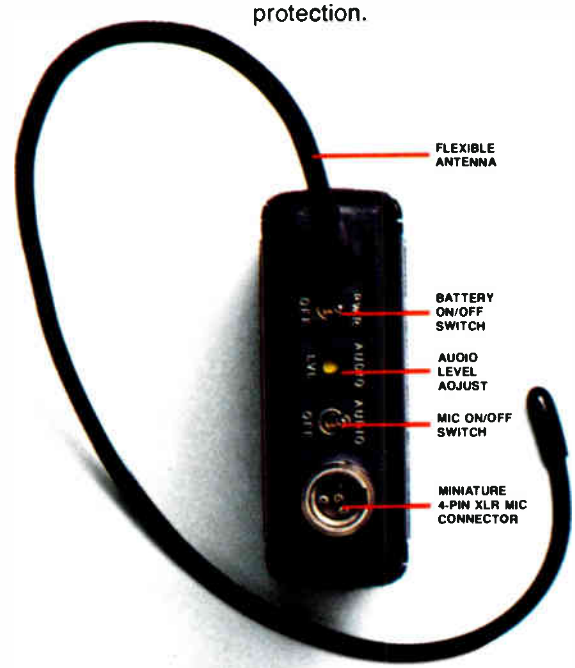
Ranger T-94 and T-99A handheld transmitters

Ranger offers two handheld transmitters that provide excellent audio performance via CVX audio processing.

The T-94 is a condenser cardioid transmitter featuring Vega's exclusive K-4 element. The T-99A is a dynamic cardioid transmitter featuring Electro-Voice's renowned N/D757 N/DYM® element, which delivers higher sensitivity, greater vocal range and better intelligibility than traditional vocal microphones.

A microphone level adjustment optimizes system performance with different performers. A silent on/off audio mute switch allows the system to be turned on and off while the RF remains on.

The T-94 and T-99A deliver a full 50 mW of RF power via patented* internal dipole antennas for high antenna efficiency and maximum operating range. Each features a molded ABS case with internal webbing for rugged protection.



Ranger T-93 bodypack transmitter

Ranger T-99A handheld transmitt

— omni
:M-44B)

LM-202X — omni
(Crown GLM-100/E)

113X — omni
(Shure 838)

Ranger Specifications

Overall Ranger System Performance

Frequency Stability:	±0.005%
Working Range:	Up to 1,000 ft. under ideal working conditions; usually somewhat less in typical applications
Frequency Response:	60 Hz to 14 kHz, ±1.5 dB
Harmonic Distortion:	0.4% maximum, below transmitter limiting; 0.25% typical at 1 kHz
Ultimate S/N:	102 dB (flat) minimum (105 dB typical A-weighted)
Operating Temperature:	−20°C to +50°C (−4°F to +122°F)
Audio Output, Line Level:	+10 to −18 dBm (at full deviation), +16 dBm minimum at clipping
Mic Level:	−18 to −46 dBm at full deviation (−30 to −60 dBm with normal headroom)
Adjustments and Controls:	Power switch, mic/line switch, output level adjust control
Power Requirements:	110-130 V ac (with supplied UL-listed wall-type power supply), or +13 to +18 V dc (with Switchcraft No. 760 dc plugs)
Power Consumption:	<5 W
Dimensions, R-98:	18.1 cm (7.1 in.) wide, 20.8 cm (8.2 in.) deep, 4.5 cm (1.75 in.) high
R-97:	13.7 cm (5.4 in.) wide, 17.8 cm (7.0 in.) deep, 4.5 cm (1.8 in.) high
Weight, R-98:	1.0 kg (2.2 lb)
R-97:	0.63 kg (1.4 lb)

Models T-93 Ranger Transmitter and T-94/T-99A Ranger Handheld Transmitters

Power Output:	50 mW nominal
Microphone, T-93:	Practically all lavalier electret mics
T-94:	Vega K4 cardioid electret (condenser)
T-99A:	Electro-Voice N/D757 N/DYM® cardioid dynamic
Controls:	Power on/off, audio level adjust, mic on/off
Battery:	One 9-V alkaline
Battery Life:	8-10 hours
Dimensions, T-93:	9.7 cm (3.8 in.) high, 7.1 cm (2.8 in.) wide, 2.5 cm (1.0 in.) thick
T-94:	27.7 cm (10.9 in.) long
T-99A:	26.4 cm (10.4 in.) long
Weight (including battery), T-93:	180 g (6 oz)
T-94:	270 g (9.5 oz)
T-99A:	290 g (10.2 oz)

*U.S. Patent #4,344,184 issued August 10, 1982.

WARRANTY

All Vega RANGER products are guaranteed against malfunction due to defects in materials and workmanship for a period of 3 years, beginning at the date of original purchase. If such a malfunction occurs, the product will be repaired or replaced (at our option) without charge during the period and under the limitations stipulated in the data sheet or owner's manual (if any) for that individual product. If delivered prepaid to the Vega factory or an authorized warranty service center, the unit will be returned prepaid. Warranty does not extend to finish, appearance items, or malfunction due to abuse or operation under other than specified conditions, nor does it extend to incidental or consequential damages. Some states do not the allow the exclusion or limitation of incidental or consequential damages, so the above exclusion may not apply to you. Repair by other than Vega or its authorized service agencies will void this guarantee. Information on authorized service agencies is available from the Vega factory.

Vega
a MARK IV company

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Buchanan, MI 49107
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FAX: 616-695-1304

(continued from page 36)
telligibility (SI) are predictions not used as commonly as Alcons. The program is for evaluation and modeling of open and confined speaking and listening spaces. Voice levels, distance, sound amplification or attenuation (obstructions), and ambient noise levels are taken into account to predict AI and SI. The measure of communication effectiveness is AI, which is defined and determined by ANSI standards. AI is computed in full or 1/3 octave bands. SI is derived from curve fitting SI versus AI curves.

Measured, known, or predicted values of voice levels, source and receiver room constants, ambient levels, transmission loss, noise reduction and artificial source and receiver room levels are all user specifiable.

Five values of estimated source or receiver room quality are considered: dead, medium dead, average, medium live and live. Associated with each quality is a room constant (average absorption coefficient) which is assumed to be valid at 500 Hz and represents the average for all frequencies.

The program runs on IBM compatibles and would be handy for Umbulus users.

TEST EQUIPMENT THAT MEASURES INTELLIGIBILITY

The purpose of the B & K RASTI meter system is to function as a simple measurement device to predict speech intelligibility.

The RASTI method is based on the idea that intelligibility is determined by accurate transmission to the listener of certain well defined aspects of human speech. A transmitter produces an artificial speech-like signal that is picked up by a receiver at some listening position. The received signal is compared to the known characteristics of the transmitted signal, and the score value is calculated.

Alternatively, later work by B & K demonstrated RASTI using a dual channel FFT spectrum analyzer. For now, this approach is quite awkward and not directly supported by any dual channel analyzers.

TECHRON TEF 12

In 1987 a software package for the TEF System 12 Analyzer was developed that does speech intelligibility measurements. Aside from Alcons, the software can perform a complete STI measurement over the range of 125 Hz to 8 Khz in two minutes. It can also measure a TDS equivalent of the RASTI method by testing only at the 500 Hz and 2 Khz octaves. Limiting the test to two octaves yields a quicker test time of 30 seconds.

The full STI test is accomplished by measuring seven individual one second time span ETC's at each of the octave center frequencies between 125 Hz and 8 Khz. After each ETC test, the MTF is calculated and the STI in each octave band is computed. Each ETC test is designed to simultaneously measure the background noise in each octave band so that the noise is automatically taken into account in the MTF. The TEF test generator level at each octave band is adjusted to match the average spectral content of speech. At the conclusion of the test, the overall STI value is computed by taking a weighted average of the individual octave band STI values.

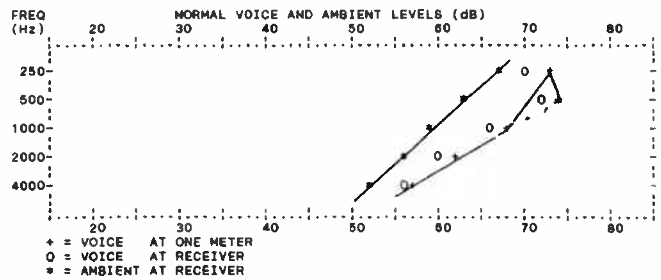
The TEF RASTI equivalent measurement gives equal weights to the 500 Hz and 2 Khz octave bands which is in agreement with the weighting used in B & K's RASTI technique. The TEF RASTI

*** ARTICULATION INDEX (AI) AND SPEECH INTELLIGIBILITY (SI) **

PROJECT ID: EXAMPLE 0
FILENAME FOR INPUT: EXO
SOUND FREQUENCY: 5 FULL OCTAVE BANDS
VOICE LEVELS: IDEAL
SOURCE ROOM CONSTANTS: KNOWN
AMBIENT LEVELS: MEASURED
NOISE REDUCTION: SOURCE ROOM QUALITY ONLY
DISTANCE TO LISTENER (ft): 6

NORMAL VOICE

FREQUENCY (Hz)	VOICE ● ONE METER (dB)	VOICE ● RECEIVER (dB)	AMBIENT ● RECEIVER (dB)	NOISE REDUCTION (dB)	SOURCE ROOM CNST (sqft)
250	72.5	70.0	67.0	2.5	2029
500	74.0	72.2	63.0	1.8	1526
1000	68.0	66.4	59.0	1.6	1367
2000	62.0	60.4	56.0	1.6	1373
4000	57.0	55.9	52.0	1.1	1152



VOICE QUALITY	RATING AI	SISK (%)	SISF (%)
NORMAL	0.18	71.8	44.7
RAISED	0.38	96.1	91.6
LOUD	0.58	98.1	97.8

TPM Software, Articulation Index and Speech Intelligibility.

measurement method used here also makes use of all the one- third octave modulation data from .5 to 12.5 Khz for both octave bands. This is in contrast with the conventional RASTI technique, as implemented by the B & K unit, which only uses the modulation data from 4 one third octaves bands at 500 Hz and 5 bands at 2 Khz. This spacing of modulation data means that in some situations incorrect intelligibility numbers will be predicted. The software also provides for complete raw data storage and retrieval of all test data. This allows a freedom in future post processing of the data to yield new information.

DRA LABORATORIES MLSSA

MLSSA (Maximum Length Sequences System Analyzer) is an analyzer which

(continued on page 65)

Modulation Transfer Function Matrix

Frequency Hertz	1 125	2 250	3 500	4 1000	5 2000	6 4000	7 8000
0.71					0.930		
1.00			0.828				
1.41					0.849		
2.00			0.730				
2.80					0.791		
4.00			0.639				
5.60					0.781		
8.00			0.571				
11.20					0.711		
16.00			0.471				

RASTI value= 0.824 ALcons= 2% Rating= EXCELLENT

ESC to exit or PrtScr to print. MLSSA: Speech Transmission Index

RASTI taken at Wolf Trap seat 0-48. Center cluster on, all else off. Courtesy DRA Laboratories.

BEARSVILLE THEATER

BY RICH GRULA

It seemed like a simple idea in 1974. Albert Grossman, legendary in the music world for the management and development of superstars like Bob Dylan, the Band, and Janis Joplin, wanted to build a theater/supper club at his Bearsville studio and restaurant complex near Woodstock, New York. A small barn was chosen for the renovation and Grossman put some of the country's top young architects and sound engineers to work on the project.

Call it a Woodstock state of mind, but there were some delays. And changes. And weird happenings. The structure was 85 percent renovated at a cost of 750,000 1974 dollars, then used only as a work space (no doubt the most expensive workshop in that country). A customized 120-speaker ceiling system was installed, but never turned on. Albert Grossman, after reviving plans for the theater in 1986, suffered a heart attack and died in a plane over the Atlantic Ocean. Now, finally, 15 years after its start, the Bearsville Theater is finally nearing completion. Mind you, it remains a few hundred thousand dollars short of being totally finished, but a light is beginning to shine at the end of the tunnel.

THE STORY BEGINS

To understand this project, one must first understand Albert Grossman. According to architect John Storyk, designer of numerous music and media facilities, Albert Grossman had two interests in addition to music — food and construction. The Bearsville Theater was the perfect combination of all three — a fabulous restaurant where you could see a great performance in the perfect structure.

Grossman's plan called for the state-of-the-art facility to be cleverly hidden beneath the barnish exterior that covers all Bearsville structures. It would include a small but comfortable theater with full lighting, sound and 35mm projection capabilities and an attached supper club separated by glass, with full view of the stage and its own top quality sound system.

Architect Storyk oversaw the initial phase of construction as the 1,800-square-foot barn was expanded to 15,000 square feet (though it would have been easier just to start from scratch, Grossman wanted to respect the site of the original redwood barn). All the walls were moved back, a basement was dug underneath, and the roof was raised two feet.

"If you stand in the theater and look at the stage," says Storyk, "from the light trough up is literally the only thing remaining of the original barn. Everything else is fake. It's basically no longer a wood barn. It's now a steel building with steel columns holding up everything."

Flexibility was a key requirement, since Grossman had grand production plans, ranging from film, to musical concerts both on stage and in various locations within the room, to boxing (Grossman owned a piece of a boxer at the time). For that reason, the room's design avoided the typical pie-shaped auditorium mold. And the seating floor, typically angled upward for better sight-lines, was left flat.

Audio requirements of the facility were handled by Ted Rothstein, then chief engineer at Bearsville Studio. For the supper club, he designed a 120-speaker ceiling system totally hidden behind pine slats that covered the entire ceiling. The full range 4½-foot speakers used, designed by Rothstein for his line of ROR studio monitors, were mounted on several heavy wood panels, which could be lowered, and wired in four groups.

"I'm not sure I would do this exactly the same way today," chuckles Rothstein. "The sound has to pass through a short distance, maybe an 1½-inch channel between the slats, before coming out. I knew it was kind of an odd way to mount them, but Albert really liked the look and he wanted that. And John thought it was reasonable idea too."

TAKE TEN...YEARS

Unfortunately, Rothstein never got a chance to test his speaker system. Just as

Circle 216 on Reader Response Card

the wiring for the ceiling system was completed, the entire project was halted by Grossman for what Storyk calls "personal, Albert Grossman reasons." Electricians scheduled to wire the speakers groups to the amplifiers were canceled and Rothstein's ceiling system was never even turned on.

Grossman, with literally dozens of other projects in the works, dropped the theater from his list of priorities. Since it was 85 percent complete and already being used as its own workshop, the theater became the central workshop and storage area for all Bearsville projects. Changes were made here and there over the next 10 years — walls moved, rewiring done — all to facilitate the space as a workshop.

In late '85/early '86, Grossman discussed plans to complete the theater with Storyk. But before anything could be done, Grossman suffered a fatal heart attack over the Atlantic on his way to England. As could be expected with an estate the size of Grossman's, complete chaos reigned after his death. Funding for projects such as the Bearsville Theater was held up for several years as lawyers sorted out the mess.

STARTING OVER

In late 1988, enough elements of Grossman's estate and trust were worked out so that funds for the theater could be allocated. When Woodstock's primary legitimate theater burned down that winter, it became apparent to all concerned that the Bearsville Theater could fill a void as the area's summer regional theater (which was never exactly at the top of Albert Grossman's list of production possibilities). Indeed, this is the only all-weather facility of any size within a 25-mile radius. Once again in the architect's chair, Storyk found much of the project's new budget consumed by changes in the world of building codes.

"When the building was done originally, there were no handicap codes, no parking lot requirements, no emergency lighting and different septic requirements," explains Storyk. "Plus there were a few changes we wanted to make because nobody had dealt with this building as a theater."

Storyk estimates $600,000 to $700,000 was spent finishing the interiors and bringing the entire premises up to code. The supper club concept was abandoned, since the Bearsville complex offers three restaurants within 50 feet of the theater. Sound contractor Rothstein was faced with the task of digging through the work of carpenters and electricians to complete his ceiling speaker system in what was now to be used as a pre-show, intermission and post-show lounge.

"It was difficult to find the original wires because so many people had worked on it over the years," he says. "They'd just cut wires because they thought it'd never be used for anything...it was a dead project. We manged to trace the wiring and patched and planned and figured out the original job enough to get it running."

Rothstein ran wiring from the four speaker groups to three old Phase Linear 700 power amps located in a rack tucked behind the stage right wall (the amps, salvaged from a Bearsville touring PA he designed in the 70s, are scheduled to be replaced by two Urei JBL 6290 power amps). After 15 years of wondering, Rothstein finally got to turn on his system last August.

"I was really impressed with it," he chuckles. "I said, "Whoa! This sounds pretty good" I couldn't believe these speakers sitting there for all those years and never turned on would sound quite that good. 15 year-old cutting [edge] technology."

Storyk's final interior plan for the theater utilized surfaces of bare pine with no special venting or traps for bass. Though some might consider such a surface too reflective for proper sound reproduction, Storyk cautions otherwise.

"Don't be deceived by wood," he says. "Yes there's a lot of wood, but nothing's parallel. It's very live and bright and the theater people love it. At 60 cycles, I acknowledge there is very little low frequency. On the other hand there is a deceptively large amount of volume for 260 people. The Bottom Line in New York City seats about 450 and I'll bet we have 50 to 70 percent more volume for half the people. If you look at RT, there are several ways to get it to work. You can either increase the volume or increase the absorption. Volume is low frequency absorption. This space is disproportionately voluminous for a relatively small population."

"Number two," he continues, "when you start to get to 400 or 500 cycles and up, a lot of rough wood is more absorptive than you think. This is pine, saw cut and very soft. If you take four-by-eight plywood and put three coats of deck enamel, that's one kind of absorption. But pine is a different animal."

"The only thing I worry about is very deep bass — full sound reinforcement, 110, full frequency down to 40 cycles. But that's not really what this space is about anymore. It would be silly to get this room to deal with that. You'd have to have a lot of pure low frequency venting. So where's it going to go? There are a lot of holes in this building — volume and physical holes. It's not like a seal studio control room. I think the volume and looseness of the building will pick up more than you'd think. If Albert were alive and this space was more for concerts, then we'd have 8-foot deep panels buried somewhere disguised with burlap. But given the nature of the building now, I think that's unnecessary. It has to be balanced for different applications."

Rothstein wired several permanent access panels for the house sound system, which is designed to be operated from several locations within the room. In the front and back of the theater, panels contain 40 mic inputs, eight line level/return lines and two intercom lines. Another panel, on the theater wall opposite the lounge area, includes an identical setup but with only 16 mic lines are located in the projection booth and in the manager's office. The central patching area is near the amp racks located on stage right.

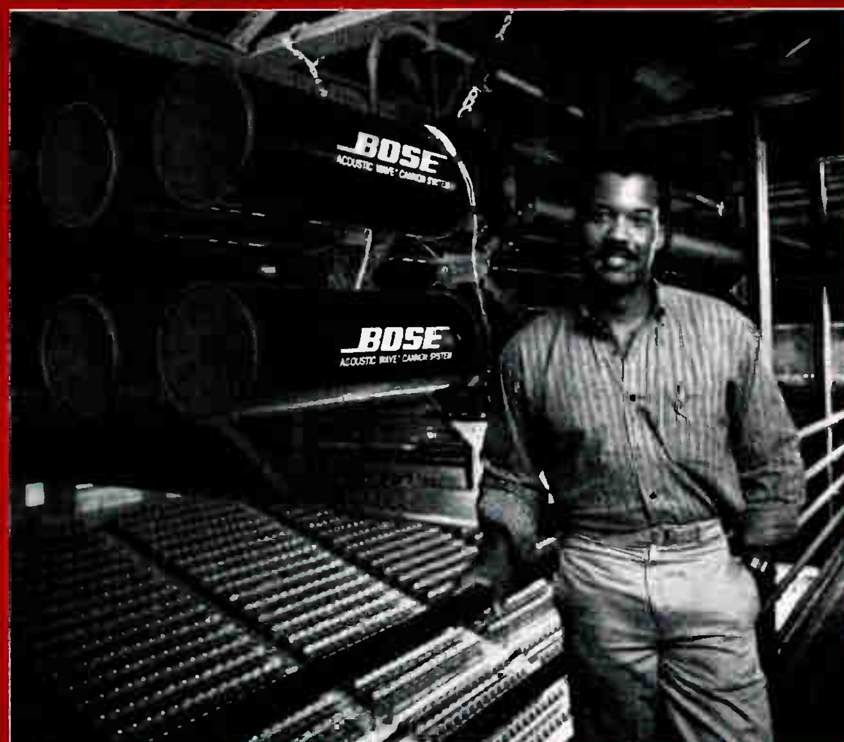
According the Rothstein, the theater's

(continued on page 71)

Bose® ProFiles

"When I went after this job, I looked at several manufacturers. But when I examined cost vs. performance and considered ease of installation, Bose was the clear choice."

—Phil Thorne
Taft Broadcasting Company



The Site: Health & Physical Education Building
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The Challenge: Provide a quality sound installation for 12,000 seat enclosed basketball arena with anticipated crowd noise level exceeding 95dB

The Contractor: Phil Thorne—Taft Broadcasting Company

The Tools: Bose® Sound System® Software

The Products: Bose 802®-II/Acoustic Wave® Cannon Central Speaker Cluster
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SOFTWARE REVIEW

THE PHD PROGRAM, RELEASE 4.0 PART II

BY MIKE KLASCO

Last month we began our review of The PHD Program and discussed its development from a design tool based on a plastic sphere and the HP 41 calculator to the present release which is fully implemented on IBM compatible/MS-DOS computers. Also examined was the comprehensive reverberation prediction utilities, spherical mapping technique, speaker aiming, and the program's approach to graphics and data reports.

This month we will explore the program's capabilities in adjusting and balancing speaker levels; performance simulations, including intelligibility and uniformity of coverage; and the Q-Plus speaker directional files utility which is bundled with The PHD Program.

POWER ANALYSIS

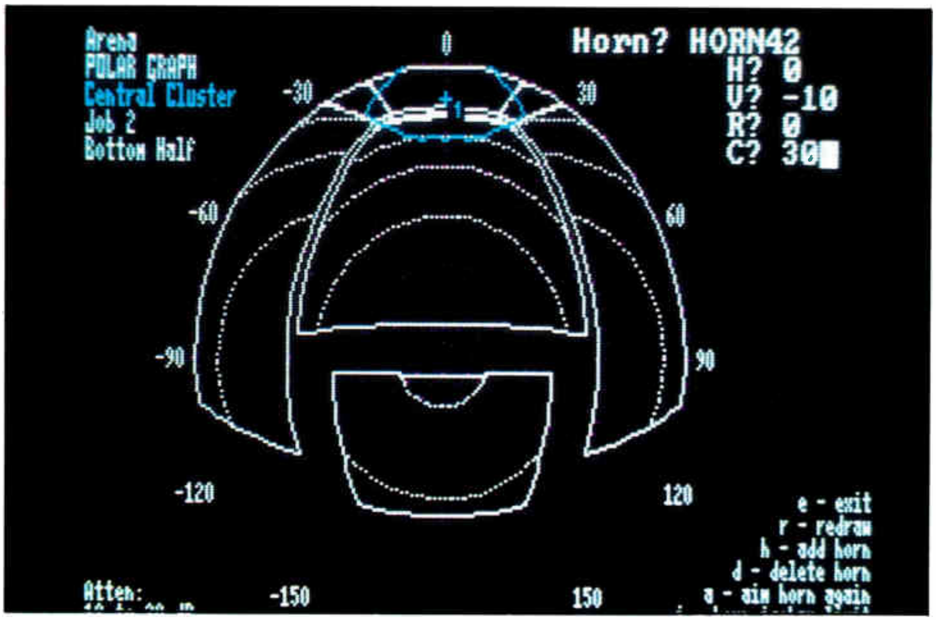
This section's objective is to calculate the required electrical power for each horn and driver combination in the Cluster. The function of this module is to achieve a desired SPL on axis in the listening areas. Different loudspeakers produce different levels with the same wattage at the same reference distance so each type of horn-driver combination must be considered individually. If the specified SPL is beyond the power rating of the compression driver, the program will indicate this.

The user is asked what sound pressure level is required plus headroom. This SPL is the average SPL so you need to add some margin for headroom. The user is also asked if the calculations will take into account the effect of direct sound absorption. This extra step would be wise when you suspect intelligibility performance might be marginal.

PERFORMANCE ANALYSIS

The Power Analysis section shows the power required for the Horn/Driver combination to achieve the desired SPL on-axis only. The Performance Analysis section shows how uniform the direct sound coverage will be both on and off axis. The interaction of the Horn/Drivers is examined in terms of Intelligibility and Acoustical Gain.

As with most of the other sound system design programs, both coherent and incoherent addition options are provided. Two phase-coherent sine wave signals of equal amplitude will combine to give a level 6 dB higher than either sine wave. This is Coherent Addition. In practice many loudspeakers are not phase-coherent with respect to each other and would add incoherently (a 3 dB increase). When adding levels from this section, you can choose Coherent or Incoherent Addition. Coherent Addition should only be considered when you have a cluster that is in time alignment (by actual on-site measurement) and furthermore only when using the same series horns. Incoherent Addition should be used for split clusters and clusters that are not super accurately aligned.



Isobar contour of a horn (blue) generated for Q-Plus over mapped room (white).

INTELLIGIBILITY

The signal-to-noise ratio (ambient noise) can significantly influence the intelligibility. This noise level is weighted as to the frequency and bandwidth most affecting intelligibility. An easy way suggested by the program (in the tutor mode) to approximate this weighting is to measure the noise level while using a weighting scale of a sound pressure level meter. You can enter the noise level once if it is uniform or separately later if the level varies from place to place, or from time to time. By entering these data separately, you will

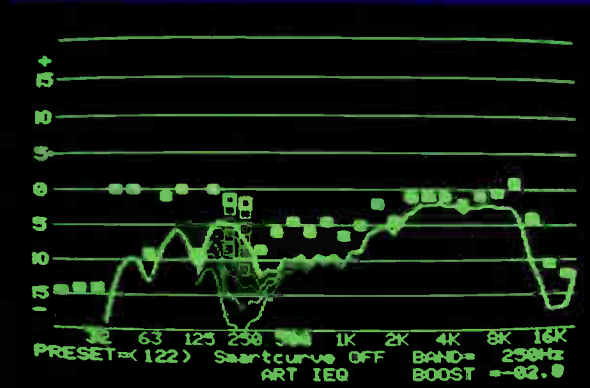
THE
IEQ
WITH
Smartcurve™

Not Just Another Programmable Equalizer With MIDI™

The IEQ with Smartcurve™ is a programmable, high performance graphic equalizer that includes a video output. For those who wish to enjoy the video output of the IEQ, ART makes the IEQ Video Monitor.* Smartcurve™, proprietary software developed by ART gives you instant actual frequency response as easy as the push of a button.
The IEQ Family consists of both the 2/3 octave and the 1/3 octave graphic equalizers. Both types come in two varieties, Controllers and Satellites. A Controller is a self-contained programmable intelligent graphic equalizer capable of controlling 15 satellites at once. IEQ Satellites are exactly the same unit except the front panel controls are eliminated.

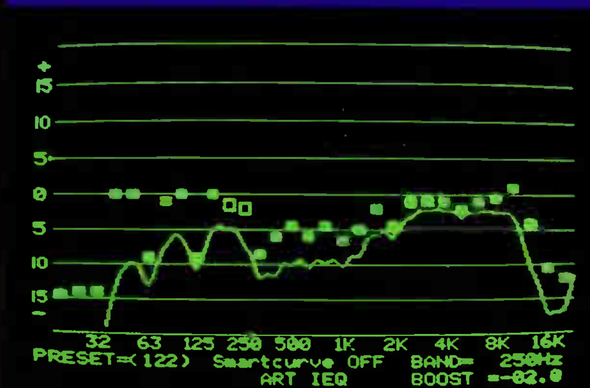
IEQ Model Specifications:
Controller & Satellite
- 128 battery backed presets
- MIDI
- Frequency Response 20Hz-20kHz ± 0.5dB
- THD - ≤ .009% @ 1kHz, 0dBM typical
- Dynamic Range - ≥ 100dB typical
- Balanced inputs and outputs

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



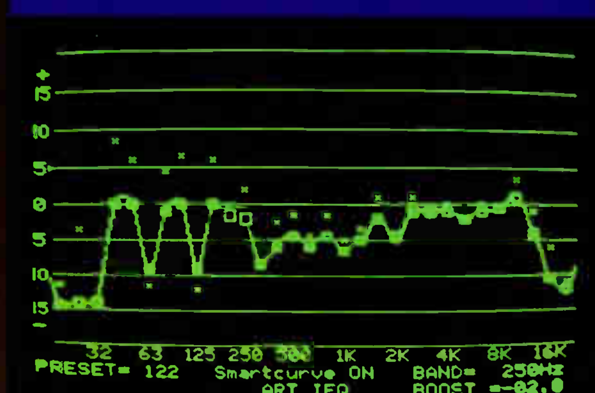
1 See the Sound

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



2 Hear the Sound

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



3 Perfect Sound

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

It Has To Be A Work Of . . .

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Circle 204 on Reader Response Card

```
CENTRAL CLUSTER --
Desired SPL: 100 dB  /  Freq. Band Centered at: 2000 Hz
-----------------------------------------------------------------
NO. HORN / DRIVER  ELECT. POWER   ROOM       ORIENTATION
    COMBINATION    (watts) from   CONTOUR   (HA/VA/Rotation)
                   LD    LD+LR    (dB)        (degrees)
-----------------------------------------------------------------
 1  H123 / 1234    13.2    6.3    24.0        0 /-25 / 0
 2  H456 / 1234     5.2    3.1    18.0        0 /-60 / 0
 3  H789 / 1234     5.2    3.1    18.0        0 /-90 / 0
-----------------------------------------------------------------
Total Power Needed:     23.7   12.4
Maximum LR** (dB):     105.6  102.7
    ** IF A TRUE REVERBERANT FIELD IS PRESENT!
     Do you want to make any changes (Y/N? n
```

Power analysis report.

have an individual chart of results for each location.

As was discussed in last month's review, the use of data reports rather than a numerical graphic plot is a carry over from the days when The PHD Program utilized older PC and TEF architecture. Aside from calculated results for each specified location, an overall report called Predicted Performance is generated. In Predicted Performance, the highest and lowest and statistical variation of direct sound coverage and intelligibility (RASTI and Alcons) are provided. By examining the data for key locations individually (from the ideal to worst case) — obstructions, overlapping horns, etc. — and also reviewing the overall performance report, the user can obtain all the information needed. Alcons and RASTI are shown on the chart for LD and LD + LR methods. "LD" is Level Direct and "LR" is Level Reverberant. Total direct SPL is also listed.

Intelligibility predictions have been a point of controversy, not just in computer software, but on a formula level. This month's feature article on intelligibility explores the intelligibility question, compares various approaches to predicting and measurement, and discusses some future developments.

The original Peutz (Alcons) formula was proven accurate with actual subject testing, but was only written for single omni-directional sources. Derivations of this Peutz formula have attempted to account for directivity and multiple sources, but these have not been conclusively tested. The Alcons formula is being widely questioned as an acceptably accurate estimation, but is still commonly used and has been included in The PHD Program (as well as in JBL's CADP and Altec's AcoustaCAD). However, The PHD Program makes use of an advanced information index technique to achieve greater accuracy instead of the more common implementation of the Alcon's method. In The PHD Program S/N ambient noise level is definable, which avoids one source of error. If the noise level varies over the listening area, then S/N can be specified for each point.

The PHD Program also provides a transformation of Alcons prediction to a RASTI score. While RASTI (which is a measurement technique, not a prediction algorithm) can be shown to be accurate and can be transformed into accurate Alcons data, questionable Alcons data cannot be turned into accurate RASTI data by being passed through a lookup table.

The results of the Performance Analysis can be saved and later retrieved and edited.

MECHANICAL DESIGN

The PHD Program does not provide a mechanical design program to aid in the design and construction of the speaker cluster. Usually some sort of drafting or

```
CENTRAL CLUSTER
-----------------------------------------------------------------
NO. HORN / DRIVER  ELECT. POWER   ROOM       ORIENTATION
    COMBINATION    (watts) from   CONTOUR   (HA/VA/Rotation)
                   LD    LD+LR    (dB)        (degrees)
-----------------------------------------------------------------
 1  H123 / 1234    13.2    6.3    24.0        0 /-25 / 0
 2  H456 / 1234     5.2    3.1    18.0        0 /-60 / 0
 3  H789 / 1234     5.2    3.1    18.0        0 /-90 / 0
-----------------------------------------------------------------
POINT 1
Room Attenuation Contour (Enter 0 to exit)? 27
 Location:
    H Angle? 0
    V Angle? 16
 Horn/Driver Number (Enter 0 if there are no more)? 1
  Horn Attenuation Contour? 3
 Horn/Driver Number (Enter 0 if there are no more)? 0
```

```
                          POINT 1

Room Attenuation Contour:  27.0 dB
Location:  0/16
                                      USING        USING
                                   LD METHOD  LD+LR METHOD
 ____________________________________________________________
|                                 |            |             |
|TOTAL DIRECT SPL:                |  94.0 dB   |   90.8 dB   |
| Variance from DESIRED SPL:      |  -6.0 dB   |   -9.2 dB   |
|ARTICULATION LOSS (ALcons):      |  10.5 %    |   10.8 %    |
|RASTI Speech Transmission Index: |  0.51      |   0.51      |
|_________________________________|____________|_____________|

       Is this point satisfactory (Y/N)?
       (If NOT - then it can be re-entered.)? y
```

Performance analysis for a single location.

sketch function is provided in sound system design programs to check for collisions between speaker components, specify rigging and interlinks between horns, create working drawings for the cluster frame, aid in the visualization of how the speakers will appear within the room, determine the cluster center of gravity, calculate the weight of the cluster, and so on. The program's developers, however, feel mechanical drawings should be done instead within one of the sophisticated CAD drafting programs already on the market. Their philosophy is "Why drive up costs which must be passed on to the user by reinventing the wheel?" If the user establishes a library of horns, the loudspeaker coordinates can be entered directly into these programs. Users can prepare the mechanical drawings manually, or on a computer with AutoCAD, Versa CAD, or Generic CAD. The program does provide data for determining exact location of the loudspeaker and aiming points in the architecture. The relative dB gives a convenient way of verifying the power ratios at the loudspeaker location.

If you choose to power the cluster(s) using the levels
found by considering ONLY Direct Sound then:

UNIFORMITY of Direct Sound Coverage	+/- 3.1 dB
LOWEST Direct Sound SPL	94.0 dB
HIGHEST Direct Sound SPL	100.3 dB
GREATEST Variance	6.3 dB
LEAST Articulation Loss (ALcons)	7.7 %
GREATEST Articulation Loss (ALcons)	10.5 %
LEAST RASTI Speech Transmission index......	0.51
GREATEST RASTI Speech Transmission index	0.57

If you choose to power the cluster(s) using the levels
found by considering BOTH Direct and Reverberant Sound then:

UNIFORMITY of Direct Sound Coverage	+/- 3.2 dB
LOWEST Direct Sound SPL	90.8 dB
HIGHEST Direct Sound SPL	97.1 dB
GREATEST Variance	6.3 dB
LEAST Articulation Loss (ALcons)	8.0 %
GREATEST Articulation Loss (ALcons)	10.8 %
LEAST RASTI Speech Transmission index......	0.51
GREATEST RASTI Speech Transmission index	0.56

Is this satisfactory (Y) or would you like to change the
Loudspeaker Cluster (N)? **y**

Performance analysis for the job.

SYSTEM DIAGRAM AND BILL OF MATERIALS

PHD generates an extensive report on the coordinates of the seating plane(s) and loudspeakers. (I also provide the prints that were marked up when the job was prepared for the data entry for the work sheets.) The loudspeaker report denotes the coordinates, aiming, model numbers, and relative levels of each of the speakers in the cluster and can serve as the basis of the bill of materials.

PHD does not provide any utilities for generating system flow charts. Various flow charting programs are available; one of the most complete and efficient MS-DOS programs I have come across is Flow Charting 11 + by Patton and Patton. Cost is $220. Flow charts can also be prepared within drafting programs.

```
Axial directivity factor (Q) .....   69.43
Directivity index (DI) ...........   18.42 dB
Energy Distribution:
  3 dB Horn Contour ..............   34.0 %
  6 dB Horn Contour ..............   21.7 %
  9 dB Horn Contour ..............   15.3 %
  12 dB Horn Contour .............   10.8 %
  15 dB Horn Contour .............    6.6 %
  Approximate 20 dB Contour ......    5.4 %
  Remaining energy ...............    6.2 %
       (To 180 degree  66  dB Contour)


            Press RETURN to continue.
```

Q-Plus data report.

Q-PLUS FOR Q AND ENERGY DISTRIBUTION

Q-Plus is actually a separate program and does not rely on data from the main software, so it can be used as a stand alone utility. The Q information can be used in the Horn and Driver Selection File of the main program, and the energy distribution data can be used in its Power Selection module whenever greater intelligibility prediction accuracy is critical.

Two choices are available for determining Q; the Angle Method and the Contour Method. The Angle Method is for users who have their own test instrumentation and desire to make their own measurements, or who have access to manufacturer's numerical data. The second option is the Contour Method, which relies on the manufacturer's published contours.

Up to the release of 3.2, the data file library of The PHD Program used the Contour Method, that is, the Contour data as it was published by the manufacturer. With the current release, many of the files are now from the latest actual computer test data generated by the manufacturers. This new horn and driver information is much more accurate.

The percentages refer to the portion of the total energy contained in each contour band. This information is useful when you have been using the main program and are at the stage of considering pattern coverage. Simply knowing the -6 dB down points of a horn (i.e. 90x60) is not enough to really characterize a horn. Some horns have soft patterns, such as with acoustic lens where energy only slowly continues to drop off after the -6 dB contour. This is useful for short throw applications with listening areas that are variable, and with rooms with absorptive surfaces. Conversely, hard pattern horns, such as the Altec Mantaray, put almost all their energy into their rated pattern, with rapid energy drop off beyond this point. This is useful for high speech intelligibility in reverberant spaces. One of the purposes of Q-Plus is to aid the program user in determining the Q, or hardness of the pattern of the horns the designer is considering for the job. Of course other considerations are also required in horn selection, such as crossover point and driver mounting requirements (1-inch, 1.4-inch, 2-inch) which have implications for maximum acoustic output, and even other obscure factors as sound quality.

CONCLUSIONS

The program is easy to learn, and the tutor mode is helpful. The layered menu system is workable, but the special function keys of the IBM style keyboard were not taken advantage of (such as using F1 for help, F2 for saving the data, F3 for printing, F4 for disk directory, etc.) A pull down menu system, such as X Windows, would be an even greater improvement.

	*** HORN/DRIVER MENU ***
(P)	TURN ON Printing
(S)	Return to Start-Up Menu
(E)	EDIT, VIEW, or ADD to Horn/Driver Data Files
(0)	Display names of ALL horns and drivers
(1)	Display only ALTEC horns and drivers
(2)	Display only COMMUNITY horns and drivers
(3)	Display only ELECTRO-VOICE horns and drivers
(4)	Display only JBL horns and drivers
(5)	Display only RAULAND-BORG horns and drivers
(6)	Display only RENKUS-HEINZ horns and drivers
(7)	Display only TAD horns and drivers
(8)	Display only UNIVERSITY horns and drivers
(9)	Display only ALTEC SYS. horns and drivers
(10)	Display only E.V. SYS. horns and drivers
(11)	Display only JBL SYS. horns and drivers
(12)	Display only R.H. SYS. horns and drivers
	Enter your choice, then press RETURN.

New organization of release 4.0 menu for the expanded loudspeaker library.

As the program is from an independent developer, the product library is more extensive than most of the other programs. The program is comprehensive in its performance predictions, including PAG-NAG for acoustic feedback, Alcons for intelligibility, and a full range of choices for reverberation.

There is some question as to whether Alcons and PAG-NAG are adequate to predict results of complex sound system designs, but for now no other solution has been offered by other programs (RASTI and STI are measurement, not predictive techniques). Some other software developers have opted to leave out intelligibility and feedback predictions until more acceptable formulas are developed. It is my preference to have some form of prediction capability, even if flawed, as long as suitable caveats are provided. The PHD Program provides these warnings both in the manual and within the program. Actually, the program suggests that the Performance Prediction not be left with the client, as it might be inadvertently taken for a performance guarantee. (See discussion of these by John Prohs.) Aside from gain before feedback and intelligibility predictions, the program also shows shadowing effects and provides for the easy relocation of clusters without data reentry. Most of these features are not found in competitive programs, or at least not within the same program.

While the predictions are comprehensive and contain useful information, the report format is an efficient but generally not intuitively satisfying solution to the communication of the results of the performance simulations. And, there is no mechanical drawing or flow chart to present, unless prepared in a drafting program or manually. Futhermore, the lack of slick graphics detracts from the program's usefulness as a sales tool, and the spherical plotting graphics might tend to be confusing to customers.

Conversely, the spherical mapping module, once you get comfortable with the "fish eye" array's perspective, is a highly intuitive path for selection and aiming of speakers. The monetary outlay of The PHD Program is half the cost, or less, of other competing programs; furthermore, the money goes to a worthwhile cause. Version 4.0 is sent to contributors of $300.00 or more to the Heyser Scholarship Loan Fund which helps needy and deserving students pursue an audio engineering graduate degree. All in all, this is a cost-effective choice as an engineering tool, but this product has limited utility as a sales presentation aide. ■

ON REPORTS AND PRESENTATIONS TO A CLIENT

By John Prohs

The client can be given data from the report which shows a very thorough design procedure was used to accomplish the system design.

The printout from the acoustical analysis shows the input and the results obtained for the acoustical analysis of the building. Along with this report one can include comments showing the acoustical flaws of the space and if acoustical treatment could be of benefit. Further description of this benefit can be tied into the polar map presentation, which shows where the sound from the loudspeaker system will go.

The polar map printout, showing the horns, can be used to show how well the room would be covered with sound. Examination of this graph will show how much care was taken to keep the sound off surfaces that could degrade the system performance, and how carefully overlap zones were chosen to be in the least offending areas (such as aisles).

The program can be used to show how a competitor's design may be flawed. A copy of the job is made, the system design is deleted, and the system specified by the competitor is entered. Errors in coverage (overcoverage or undercoverage) are immediately apparent.

In other programs it may be difficult to tell if excessive overlapping was used. The sum of these overlapped devices may appear to give good coverage, but excessive overlapping causes interference which can degrade the system performance and increase the cost of the system. The ability to see each individual pattern helps you see if there are too many, too few, or improperly aimed components.

The mapping data report can be printed out to show that all signal delay times have been specified properly and to show how loudspeaker aiming points can be verified.

The power analysis can be given to the client to show how proper power levels were determined.

Finally a carefully qualified performance analysis could be left with the client to show that the coverage and gain requirements were checked out. Here again comparison at some locations might be made with a competitor's design. It is very important to stress in writing on the report that this is not a guarantee of performance but is one of the best predictive methods currently available to designers.

The AES Heyser Scholarship Fund

The Richard C. Heyser Scholarship Loan Fund has been set up to honor Dick Heyser, a highly gifted, loved and respected engineer, with a lasting memorial. The scholarship loan will financially assist promising graduate engineering students in the field who, otherwise, could not continue with their studies.

In March of 1987 Dick died, just a few months before he would have assumed the office of AES President. He not only was active in AES but he contributed greatly to the audio field through his Time Delay Spectrometry discoveries. He gave of himself on a personal level as well. Carolyn Davis, Co-Founder of Synergetic Audio Concepts, said of the fund, "Dick, himself, gave so much to all those with whom he came in contact, especially those just starting out; we feel Dick would have been pleased to know he is being remembered in this way."

In conjunction with this memorial, a major update of The PHD Program† will be dedicated to Dick Heyser and all the proceeds will go to the Richard C. Heyser Scholarship Loan Fund.

You may obtain a program for a donation of $300.00 or more; prior owners may upgrade for a donation of $50.00 or more. Make your check payable to the Richard C. Heyser Scholarship Loan Fund. Send to the Richard C. Heyser Scholarship Loan Fund, c/o *Sound and Communications*, 25 Willowdale Avenue, Port Washington, NY 11050.

† Trademark of Ambassador College.

(continued from page 14)

them. That's the service that they're buying. In the case of the contractors, you're actually selling them something different from that. You're providing them with a service in that you've got to get in and you've got to get out. You've got to follow their schedules. But what you're selling them is the ability to move their product, and in their case their product is the homes — their spec homes. They're looking for something to differentiate their homes from other homes in the marketplace: either to sell them for more money or to sell them faster than their competitors. So, it's a different type of a sell.

MORRISON: I think Bill Ray sees every kind of selling, because he's an independent rep and he deals with a lot of different people. Bill, what do you see some of the contractors getting into that may be a little bit off beat for them?

RAY: Well, I'd like to say something about selling, since that is what I do and it's probably the single thing that most impacts this industry right now. A few years ago there was a popular theme that we were nearing an information age. And I think that's a theme that needs to be carried through into sound contracting.

Probably the most valuable thing that you have to sell as a sound contractor is the information that you're able to give your customer: the knowledge and expertise. Unfortunately, we're seeing a trend that parallels the consumer industry in hardware. I'm a rep and I hear from all you guys about prices dropping and that I've opened too many dealers, and so on. But the reality of the situation is that as our products become easier to build, and as they're building larger quantities, they're going to become more consumer oriented and consumable.

The asset that you people have as contractors is one of expertise and knowledge. You're never going to win the battle, no matter who you're selling to, when it becomes a question of who has got the lowest price in a quantifiable object, whether that be a speaker, microphone or whatever. Where you're going to win the battle is in quality. Qualifiable service. Not quantifiable. So, if you can provide expertise, the kind of quality installation that Tim was talking about, where the customer says, "I never thought wiring was suppose to look like that," these are the things that are important. These are the things that impress people.

Regardless of the market you're addressing, whether you're putting a sound system in the Opryland Hotel, a church sound system, a house sound system or any new market, I think the underlying theme should be one of providing excellent service, excellent support. These are things that a consumer-oriented facility that sells boxes is not going to be able to compete with, and I think that should be an underlying theme for everybody in this building.

As far as new markets that I'm seeing emerging, there are always new small niche markets. Impaired hearing systems is becoming a fairly popular theme among politicians. And this would include churches, movie theaters, public arenas and theaters. This is a market that needs the expertise that a contractor can lend. Corporate meeting rooms is a growing market that requires a lot of expertise. The people putting these rooms

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SALES & MARKETING

(continued from page 56)

together generally are not budget conscious, they're very quality conscious. They're very interested in having precise and easy control. These kind of controls don't exist, often have to be custom wired, or custom engineered and designed. And this is where you can use your 15-20 years of contracting experience to walk into a facility and offer a large corporation the ability to be able to have a multi-media corporate meeting environment where even the owner of the company who has very little or no technical expertise can push several buttons and make everything happen. There are some old markets that contractors still can do quite well in, in terms of hotel lobby sound or lobby bar sound. There's still a big market in discotheque sound. This is a dangerous market if you go into it pricing products on a quantifiable basis in terms of, "Well I'll give you four speakers for $10." That's about how much you'll make. But if you go in with your expertise and show them you can do a quality sound system with minimal maintenance, there's still a market there. At least in the southeast we see a big market for that.

This concludes Part One. Part Two will appear in the next issue. ■

CONVENTION NEWS

(continued from page 24)

ramifications are legion.

According to Streicher, "The tentative abstract for the session is: The recent proliferation of the video rental industry has spawned a desire among consumers to recreate the theatrical experience in their own homes. Thus large screen video systems coupled with augmented stereo systems have evolved. Recently, several approaches to multi-channel home sound reproduction have emerged, all in the attempt to fulfill the sense of 'being there' intended by the film's sound designers. This session presents several of these methods, as well as new approaches to the implementation of multi-channel sound systems for use in live theatrical presentations, as well."

In addition to the growing 3-D market, several other categories are on the must see list. The category of Large Array Systems has, at this writing, six papers being presented. These papers focus on sound coverage, phase and dispersion, and new methods for estimation and measurement of system performance.

Another related topic deals with sound reinforcement. This category also has six papers which cover speech intelligibility analysis, loud speaker selection, design concepts for mixing consoles for multichannel reinforcement systems for music and theater, techniques for system performance optimization, and the perception of comb filtering and the use of stereo synthesis to reduce this subjective/objective interference.

Six papers are being presented on transducers and their design as regards cabinets, room effects on subwoofers, and the measurement of operating modes on a loudspeaker cabinet.

The category of New Directions includes a paper on a new kind of intercom system featuring one-digit dialing and any number of simultaneous calls. A paper on the production and application of 6-nines copper to audio [99.9999% pure—Ivory Soap, move over!] will be presented as well.

Also, there will be a workshop on fiber optics at the Hilton presented by Al Grundy from the Institute of Audio Research. This workshop will cover the basics of digital interconnect, a brief history of fiber optics and terminology, and some of the applications.

3-D sound promises to be the wave of the future, and thusly will have a major impact on the way live venue facilities are electronically and acoustically designed. The Eighty-Seventh AES convention in New York addresses these issues and truly lives up to its theme of "Audio for the Next Decade and Beyond." See you there! ■

RIVER DOWNS

(continued from page 28)

example, has two remote switches and two mic inputs. If a meeting or a banquet is taking place in the clubhouse, the remote switch to the DJ-4102 activates the local mic and shuts out the track announcer's call. But the main telephone switchboard operator can be heard in an emergency page throughout the building; it overrides all other switching. The emergency key switch activates five relays which shut out all inputs except the operator's.

An emergency page enters the system

through input 4 of the lower clubhouse mixer and it exits through output 2. Our use of the 4102's dual output here means we do not have to use an additional mixer.

WIDE COVERAGE

The sound system's basic design is by Ewing, Cole, Cherry & Parsky, a Philadelphia-based architectural firm that specializes in racetracks. Our own firm, Industrial Communication & Sound, selected the equipment.

The audio system includes nine Altec 8551 programmable equalizers (one for each of nine areas); 20 Altec 9444A dual-channel 300-watt power amplifiers; 33 TOA 380SE three-way speakers in the grandstand and an additional seven in the paddock and pavillion. Thirteen Altec MR-94B horns cover the apron (300 watts at 70 volts); 112 Altec 409 8-inch coaxial speakers cover the mezzanines and press-box; 30 B.E.S.T. 72D speakers blend into the clubhouse ceilings.

Their coverage overlaps. The grandstand speakers include mid-range horns with 90-by-40 degree coverage, the same as the horns at the apron. In the stands, the 90-by-40 degree coverage, same as the horns at the apron. In the stands, the 90 degrees is horizontal; at the apron, it is vertical, to cover a deeper area. The grandstand speakers are bolted to the beams. Two-thousand- pound-test aircraft cable is looped through each speaker cabinet and over the beam for an extra margin of safety.

DELIVERING SATISFACTION

The River Downs people wanted a sound system that their customers could hear everywhere. They also wanted fact service; we had less than three months to furnish and install the intercom, telephone wiring, computer wiring, security TX, entertainment TV, satellite dish, and sound. The system alone costs $165,000. At a racetrack, as at any other entertainment center, quality of sound is an essential element of customer satisfaction. ■

Paul Clark is project manager of Industrial Communications & Sound Company (Cincinnati, OH).

LOUISIANA

(continued from page 29)

hook the speaker lines one at a time to a special test amplifier. By monitoring the current on the output of this amplifier the computer can determine the impedance of the speaker line. This is compared to a value stored for that line to determine the condition of the speaker and line.

In three minutes, the IED computer can check the entire sound system, including speakers and print out a list of any problems. This can even be done during the night with out an operator in attendance.

Even though the original speakers and amplifiers are still in use, the quality of the sound in the Superdome is improved significantly. This is due to the cleaner front end electronics and the use of additional signal processing.

LOUISIANA SUPERDOME
PARTIAL EQUIPMENT LIST

2 IED 490RM comnputer and monitor; 4 IED 516 matrix card; 1 IED 516M mainframe; 5 IED 564 relay card; 3 IED 596 monitor card; 4 White 4400 third octave filter; 6 Audio Digital ADD-3 audio digital delay; 4 Audio Digital ADD-2 audio digital delay; 5 Aphex 301 Compellor; 2 Aphex 303 Compellor/Aural Exciter; 2 Aphex 700 studio dominator; 1 ROH 220B mainframe; 6 ROH 212B/M 1X6 DA; 4 ADC PPA3-14NO Mark II patch panel; 32 Soundolier RPM-8 Soundolier relay pak; 256 Soundolier RLM-24-5 relay; and 2 Soundolier PS24-100 power supplies. ■

Steven M. Hodge is President of S. M. Hodge Co. Inc., College Station, Texas.

DAT RECORDERS: A PRIMER

BY BARRY McKINNON

The R-DAT (Rotating head Digital Audio Tape) recorder that has come to be known simply as DAT bears more than a passing resemblance to the Beta or VHS recorder. It shares many of the same building blocks as the CD player. It uses similar technology to the PCM adaptors in use now with various video tape formats. But DAT is a new medium.

The speed at which a product like DAT would be accepted is normally limited only by market supply and demand, but DAT has been wrapped in controversy with government restrictions pending the resolution of the copy-code problem. A lot of potential users have been in a holding pattern, not getting too enthusiastic about a product that they thought might never arrive in a valuable from. But DAT for professional users is here now, and a look at the nuts and bolts of how it works and what problems it solves just might help drum up some enthusiasm.

The machines on the market provide a variety of input capabilities, and most professional machines use XLR line level inputs while most consumer machines use RCA line level inputs. The portable professional units include XLR microphone inputs, with phantom power. Being a digital audio recorder, one of the first steps is the analog to digital conversion at the input of the recorder. In all the recorders, consumer or professional, the A/D uses a sampling rate of 48 KHz, and 16-bit sampling. This translates out of computer language to mean that 2^{16} or 65,536 discrete amplitude levels that can be recorded, the quietest possible signal would have a value of 1 and the loudest possible signal should have a value of 65,536. This is done 48,000 times per second. What you have is one second of audio signal diced into 48,000 slices of time, each containing 65,536 descriptions of level or a total of 3,145,728,000 points

This digitized audio signal is then recorded to tape as a string of 1s and 0s.

This string or serial data bears no resemblance to actual audio. In fact, much like a TV picture is composed of scan lines that are stacked and interleaved to form a picture, the digital audio data needs to be reassembled again for playback. Markers in the data stream provide the reference points to stack the signal back together again, much like the bumps on Lego blocks. The reassembly is done in the digital-to-analog converter that operates in the reverse manner from the A/D . Many machines use a D/A converter that runs at a higher sampling rate than the 48 kHz A/D section. While this may seem like a waste of time, since the result provides no additional music information, it provides a worthwhile service. The higher sampling rate moves the noise generated in the decoding process up to a much higher frequency. By processing the resulting analog signal through a low pass filter, knocking off the noise components outside of the audio band, the noise can be eliminated. The low pass filter has always been a part of the D/A process, but in the past, when the same sampling rate of 48 kHz was used, an ultra-steep "brick wall" filter had to be used at 22 kHz. Ultra-steep filters ring, creating enormous overhang in the signal. This turned out to have audible consequences. By raising the sampling frequency and shifting its attendant noise, a much gentler slope filter can be used. Filter resonance is reduced and phase response is now much more linear in the high frequencies.

Panasonic's SV-250 DAT recorder.

By changing the signal from analog to digital, the quality of the recording is no longer dependent upon how accurately the storage medium (magnetic tape) can store and retrieve the voltages of the audio signal. By giving each point of the audio signal a code number that describes its voltage, all you have to be able to do is read the code number back from the storage medium (DAT cassette) and reassemble the wave form. Since the background noise of the tape and electronics do not have a code number associated with them, the reconstruction process ignores them. In this way digital audio recording gets around the limitations of dynamic range and frequency response that analog audio recorders have. The DAT system provides a flat frequency response of 5 Hz to 22,000 Hz using the 48 kHz sampling rate. With a dynamic range of 96 dB, noise reduction systems are not required. Crosstalk is a very low 85 dB. A look at a DAT recorder spec sheet might not even show numbers for wow and flutter and signal-to-noise ratio. Distortion is typically under .05 percent, low even for electronic equipment.

DAT recorders have another feature, multiple sampling rates. The consumer machines are limited to two sampling rates: 48 Khz for the analog inputs for record/playback and the CD standard of 44.1 kHz, available in playback only, for prerecorded DAT cassettes. This is a more reasonable answer to the entire copyright problem than the CBS copy-code system. A consumer DAT machine will not record from the digital output from a CD player to its own digital input, because the sampling rates are not compatible. The subcode that travels with the pre-recorded audio can include a copy inhibit that will prevent even a professional machine from recording copyright material. All professional DAT recorders such as those made by Sony, Panasonic, Nakamichi, Fostex and Tascam all will operate in record mode at the 44.1 kHz sampling rate, saving the expense of sampling rate conversion when the time comes to have a master tape turned into a CD.

One of the major markets for the pro DAT recorder is studios requiring a standards conversion system between DASH, R-DAT, CD and the satellite broadcast system. As you might have realized, the standard sampling rate of 48 kHz actually provides a wider bandwidth than is available from a CD player. You actually lose a bit of bandwidth when you go from DAT to CD, which certainly was never a problem with analog recorders.

A third sampling rate of 32 kHz for the AES/EBU digital inputs is available on some machines. The 32 kHz sampling rate is compatible with the BS-2 satellite broadcasting mode, allowing digital audio from TV satellite broadcasts to be recorded. It uses the same 16-bit linear sampling, but due to its lower sampling rate, it has a bandwidth of 15 kHz.

Digital audio tape systems are not entirely new. PCM adaptors were quite the rage for a while, until the inherent problem of muting from dropouts on the videotape or muting from overdriving the analog-to-digital converter became too frustrating. PCM adaptors have found niches in the market, but those limitations probably halted widespread acceptance. There is a PCM recorder system using 8-bit non-linear sampling at 31.5 kHz on 8mm videotape as well, but its limited bandwidth of 15 kHz prevented professional acceptance.

The analog-to-digital converter in the DAT system avoids the problem of signal muting in the presence of excess level by using a floating point sample. If the signal would demand numbers beyond the 65,536 available to describe its level, the A/D drops the lowest or least significant bits as required. This results in distortion, in what can best be described as a brittle sound, but it does not mute. This feature will be a key factor in the acceptance of the DAT in film and TV audio uses where brief instances of distortion are easier to deal with than missing information.

The tape transport in the DAT recorder is similar to a video recorder. The tape is pulled out of the cassette case and wrapped around a rotating drum with the heads on it. In most machines this drum is 30mm (just under 1.25 inches) in diameter, spinning at 2,000 RPM, and the tape is wrapped around 90 degrees, or one quarter of the drum. This is half of the tape wrap used in Beta and VHS tape, allowing faster tape handling while the tape is in contact with head drum. This translates to a search speed as high as 200 times normal playing speed. Not all DAT recorders use the 30mm drum, the miniature Panasonic SV250 recorder uses a smaller head drum, spinning faster, and a 180-degree tape wrap. This sacrifices some high-speed tape handling providing a search speed of only 60 times the playing speed, but allows an overall smaller construction. The smaller format does present some compatibility problems as we shall see. The other aspects of tape handling are impressive. Fast forward and rewind times are 40 seconds for a two-hour tape. This high-speed tape handling puts analog cassettes and open-reel recorders into the tortoise category.

The resemblance to a video transport extends to the sensitivity to moisture on the head drum. In the event the humidity sensor detects moisture on the head drum, the DAT recorder will not function until the temperature and humidity stabilize in the machine. Any condensation on the drum would cause the tape to stick and be badly damaged. This should be kept in mind when using a portable DAT recorder in colder temperatures. Don't leave setup until the last minute, bring the recorder into the building and let it come up to room temperature an hour or so before use.

Like video recorders, the actual information is recorded diagonally on the tape. This is a result of using a helical scan system. The tape has a linear travel speed of a glacial 8.15mm/second ($\frac{5}{16}$ of an inch/second), or about one half the slowest speed of an analog audio logging recorder. But by spinning the head drum at 2,000 RPM, a relative tape speed of 3.133m/second (approximately 10 feet/second) is achieved. The high effective tape speed allows 2.46M bits of information to be laid down on tape every second, for a total storage capacity of 1,200 Megabytes on a two-hour tape. That makes CD-ROM at 540 Meg and 3.5-inch disc drives at 1.4 Meg seem like lightweights. There may be applications for archiving picture information on DAT.

The actual signal is recorded in diagonal strips 13.6 microns wide. To get a feel for how wide that is, take a human hair and slice it lengthwise into five strips. One of those strips is approximately 13 microns wide. There are odd and even heads on the head drum, typically two, although some of the newer machines feature four heads to allow off tape monitoring during recording. Each of these odd and even heads lays down a stripe with the azimuth offset ± 20 degrees. The difference in azimuth of 40 degrees attenuates the adjacent data tracks significantly. Each stripe contains the odd or even information from both audio channels, the subcode and automatic track finding information as well. This interleaving of odd and even information, along with a double-encoded Reed-Solomon error correction code system (multiplexed in with the signal), gives the signal a high likelihood of survival, even in the event of a clogged head. ■

This concludes Part 1. Part 2 will continue next month.

SPEECH INTELLIGIBILITY

(continued from page 43)

makes audio and acoustic measurements based on a new approach. The analyzer is actually a plug-in circuit card for IBM compatible computers. MLS uses a kind of pseudo random noise test signal known as a maximum length sequence. This test signal has several important properties which allow recovery of the impulse response from the system output. Once the impulse response is determined, many other audio related characteristics can be derived through computer post processing, including RASTI and STI.

The STI and RASTI are computed from the measured impulse response. On a 386 machine, the MLSSA requires 2 minutes to perform an STI calculation, and a RASTI requires 10 seconds under the same conditions. Alcons is also computed. MLSSA may have some inherent advantages over the TEF system for measuring STI, but this technique is new and comparative field tests will be needed to prove this out.

Aside from STI and RASTI, the analyzer also measures frequency response, phase response, energy-time curves, reverberation time and distortion.

Another interesting feature of the DRA MLSSA that foreshadows future applications is that it can transfer its frequency response and phase measurements to Scientific Design Software's CACD crossover design program. Other possible file transfer possibilities of this and other analyzers are polar/directional data of horns and arrays being fed directly into directional files in sound system engineering software, energy-time curves being entered directly into the room model instead of ray tracing and attempting to predict the reverberation time, and so on.

ACOUSTICAL SIMULATIONS

By using ray tracing, a computer model can simulate a test signal being reflected off the room boundaries, taking into consideration the absorption coefficients of these reflecting surfaces. The arrival times and amplitudes of these reflections (at some defined location) are plotted into a facsimile of an ETC (energy-time curve). After using a test signal within the room model, music or speech could be used in this simulation and the room characteristic within the model could be superimposed on the signal. The signal would be digitized, "passed through" the room, and then converted back to audio. The "sound" of the room could then be evaluated.

This technique has been tried by a number of the large acoustical laboratories using large mainframe computers, and is sometimes part of the design process of creating new concert halls. An example of this procedure was reported in a 1984 AES preprint "Application of Acoustic Simulation To The Olympic Main Coliseum."

The MIDI interface, analog and digital signal processors, and test instrumentation based on the personal computer is bringing some of this capability down in cost. Don Keele of Techron described how a digital reverberation device was used as a part of a system to test the new STI and RASTI algorithms on the TEF 12 in a paper given at the 1988 AES International Conference on Sound Reinforcement. More recently, DSE Productions of Canada has devised a system of off-the-shelf signal processors using a digital reverberation MIDI controlled device (such as available from Roland and Yamaha), time delays, equalizers, recordings of ambient noise from the site under analysis, and a Techron TEF analyzer, but for the purpose of aurally demonstrating different reverberation times and the effect of background noise on the intelligibility.

At least one of the sound system software developers that uses ray tracing is experimenting with using the model derived time arrival data (i.e. the pseudo energy-time curve) to directly control a programmable digital reverberator through a MIDI interface. This would allow for both aurally listening to the sound of a room that has not yet been constructed as well as permitting STI and RASTI measurements.

Nippon Columbia has introduced a CD which was recorded in an anechoic chamber. This would be especially useful source material for this application, eliminating the double reverberation of the original recording and the superimposed room characteristic.

(continued on page 71)

Contracting Close-up

Birmingham International Conference Centre

Tannoy-Audix (Saffron Walden, UK) will supply sound, communication and video services in an 11-hall complex at the Birmingham International Conference Centre, due for completion in 1991. The complex will be used for a range of facilities, such as conferences, musical and theatrical performance, concert recording, and direct radio and TV broadcast via a TV studio to be constructed for use by the BBC/ITA.

Facilities will include PA for music and speech performance, recording and broadcast video links and monitoring and simultaneous translation, for the Centre's international activities.

Incorporated into the system will be computer-controlled sound distribution and switching system, together with a microprocessor-controlled amplifier called "Vector," which was developed by Tannoy-Audix for use in multi-facility complexes. Using a combined microprocessor/mic talkback unit, prioritised sound requirements can be programmed into Vector. As the system requirements expand, the amplifier can be reprogrammed to provide update services. Also, Vector's plug-in amplifier power modules can be added to, in order to increase system power.

The Birmingham Symphony Orchestra will have a permanent home in one of the halls. Tannoy-Audix will supply a system there that will allow for live performance, recording and direct broadcasting audio and monitor links.

DCI Upgrades University Stadium

Diversified Concepts, Inc. (Marcellus, NY) recently completed a renovation of the sound reinforcement system for the Syracuse University Carrier Dome Stadium.

According to Thomas A. Zorn, general manager at DCI, the new system replaces the old single-loudspeaker cluster with a split central loudspeaker cluster and a ring of 10 smaller loudspeaker clusters.

The design criteria for the original sound system was for a maximum volume of 85 decibels. The new sytem uses more than 18,000 watts of audio amplification to produce sound pressure levels up to 100 decibels. Reverberation has been reduced through closer speaker placement to the patrons. Furthermore, DCI designed the new system to sense and compensate for crowd noise.

Other additions include coverage of the uper and lower concourses that enables patrons to hear announcements in the concession and rest room areas, and a solid state signal delay and switching system to provide coverage of the seating areas during basketball games and special events.

The Conference Centre is due for completion in 1991.

People

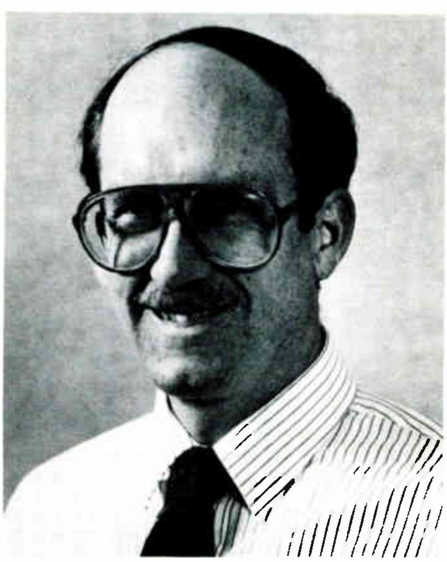
Wellikoff Now Celestion's President; Stiernberg Takes Newly Created Position

Wellikoff Now Celestion President

Peter Wellikoff has been appointed president of Celestion Industries, Inc. (Holliston, MA), its subsidiary for consumer, professional and OEM speakers in North America. A 20-year veteran of the hi-fi industry, he joined Celestion as executive vice president two and a half years ago. Under his tutelage, the company's sales have increased by about 115 percent, and distribution channels have been redefined to continue the growth curve.

Shure Appointments

Robert Gilbert has been appointed vice president, sales and finance, at Shure Brothers Inc. (Evanston, IL). In this role he will be responsible for the management of Shure financial operations and sales activity worldwide. Gilbert's previous position at Shure was vice president, finance.



Robert Gilbert

The company has also appointed four new managers to positions in product marketing: John F. Phelan, now director, technical markets, will oversee marketing efforts for Shure pro products, sound reinforcement products, and communications products. Donald S. (Sandy) Schroeder assumes responsibility for the overall management of all Shure mic product lines worldwide. He has been promoted from marketing manager, sound reinforcement products. As director, mixer products, Michael Pettersen is responsible for management of all Shure automatic and manual audio mixer product lines, as well as serving as liaison to electroacoustical consultants worldwide. Alan B. Shirley has been appointed product line manager, wired mics. In this position, he directs the daily management of all wired mic product lines.

Stiernberg Joins Infinity

John Stiernberg has been appointed to the newly created position of director of environmental products for Infinity Systems, Inc. (Chatsworth, CA). Most recently vice president of marketing for dbx, Stiernberg was also national sales manager of the professional products division for Bose Corp.

Gauss News

Paul V. Hugo has been appointed national sales manager of loudspeakers for Sun Valley, California-based Gauss. He joins the company from Audio Reinforcement Technologies Inc. (Miami, FL), where he was a co-founder and vice president.

Joe O'Connor has been promoted to customer service manager, tape duplicators. He has been at Gauss for eight years, and had been customer service engineer, tape duplicators, before this promotion.

Gerry Barclay Heads Crown Marketing Services

Crown International (Elkhart, IN) recently announced that Gerry Barclay has been appointed marketing services coordinator. In this capacity, she will be responsible for developing advertising campaigns, promotions, artist endorsements, and sales literature of Crown's electronic products line. Barclay co-owned and operated Barclay Recording and Electronics in Philadelphia.

Gerry Barclay

Kadar Now Apogee's Chief Engineer

Steve Kadar has joined the staff of Apogee Sound, Inc. (Petaluma, CA) as chief engineer. He is responsible for new product development, research, technical support and quality control. Kadar was formerly chief engineer at McCune Audio/Visual.

Sacchetti Joins Symetrix

John Saccetti, former chief engineer at Westlake Audio in Los Angeles, has joined the Symetrix Digital Processing Recorder DPR100 development team. His first assignments on the project include design of interface circuits for SMPTE, VITC, and house sync.

Mundhenk Promoted

Danny Mundhenk has been promoted to eastern region sales manager of Solid State Logic, Inc. Mundhenk has served as a salesman for SSL in its New York office for two years. In his new position, he will manage all SSL sales activities in music and film recording, post production and broadcast throughout the eastern U.S.

Danny Mundhenk

Jinks Takes New Position

Spectrum Signal Processing Inc. has added Barry Jinks to its team via the newly created position of vice president of marketing. Previously director of marketing for LSI Logic Corp., Jinks' new role covers all aspects of sales and marketing management with emphasis on sales channels and emerging market opportunities.

REP NEWS

The Texas-based firm of Jones Audio Sales was awarded "1989 Rep Firm of the Year" by Electro-Voice, Inc. The announcement was made recently at the Electro-Voice 1989 International Sales Meeting. Upon receiving the award, Chris Jones, founder of the firm, said, "This award is indeed an honor and is particularly gratifying because of all the support we received from our dealers and, of course, Electro-Voice."... Stewart Electronics (Rancho Cordova, CA) has announced the appointment of two new factory representatives to handle its direct boxes, mixers and amps. The firms are Pearson & Pearson (Denver, CO) and North Shore Marketing (Seattle, WA).

Products

New Offerings from E-V, Elektralite

Light Controller

The STL 100 Controller from Elektralite controls the feed from audio equipment to related lighting effects equipment. It features a beat frequency mode that allows the coordination of lighting equipment with music beats in a variety of ways by converting audio feed to digital signals. Up to 25 fixtures can be controlled with the STL 100.

Circle 9 on Reader Response Card

E-V Debuts

On/off switches have been introduced to Electro-Voice's N/D357S and D/D757S mics. To reduce the possibility of accidental turn-off, E-V located the switch on the mic collar, away from the handle/grip area.

The DH1Amt-16, a modification of the company's DH1A compression driver, is designed for Manifold Technology applications. Two DHAmt-16 drivers mount on an MTA-22 Manifold Technology adaptor creating the DH1A/2MT high-frequency acoustic summation system.

The FM-1202ER is a 2-way, 12-inch floor monitor, and the FM-1502ER is a 2-way, 15-inch floor monitor, new to the Extended Range line. Constructed of .75-inch Road-Wood and covered with waterproof black carpeting, the monitors are also built with handles, steel corners, and rubber feet.

The MS-1000 wireless vocal mic is constructed with the EV N/D757 capsule. This system has two separate receivers on the front end.

Circle 10 on Reader Response Card

For DJs

Numark's DM1275 stereo mixer is designed for use by entry- level DJs in clubs and bars with minimal sound mixing needs. It features two stereo input channels with phono preamps and line inputs, and a third channel with stereo line and a DJ mic input.

Circle 11 on Reader Response Card

Flexible Cable

Belden's new Brilliance broadcast product line is manufactured for difficult rack system installations, CCTV Systems, and equipment/signal connections. The bare copper conductor is insulated with polyethylene. The stranded center conductor provides flexibility and, says the manufacturer, improved flex life. Two tinned copper braid shields together provide physical coverage. The cable is jacketed with PVC compound available in five different colors.

Circle 12 on Reader Response Card

Headset Circuitry Eliminates Background Noise

The Earborne Earset II telephone headset from Starkey Telecommunications Products features patent-pending circuitry that eliminates background noise. The new signal-conditioning mic circuit is designed for confidential or noisy work environments. Included with the headset is a desktop module that allows for handset or headset use, provides a "mute" button, and controls receiver volume.

Circle 13 on Reader Response Card

A New Formula

Formula Audio, Inc. has introduced the 4400 series direct radiator line, which consists of three full-range and three sub enclosures. The trapezoidal enclosures are carpet-covered, with reticulated foam over the metal grille.

Circle 14 on Reader Response Card

Inspiration Media's New Software

Bid System 1 software is a contractors' bid preparation system from Inspiration Media. It includes bid tracking, job costing, and allows the user to build a manufacturer data base and a sales information data base. Bid System 1 also provides sale management with a reports section to track all bids within the system.

Circle 15 on Reader Response Card

For Touring, Live Sound Reinforcement

DDA has introduced two new consoles designed for live sound: the Arena Monitor and the Arena VCA

House Console. The Arena Monitor is a 16-output board with EQ an all outputs and an 18-way meter panel. The Arena VCA is an 8-group mixer that employs an additional eight VCA/mute groups for use in programming level and mutes.

Circle 16 on Reader Response Card

An Aid To Crime Reduction

Louroe Electronics' Ask 4-Kit is designed for security at cash registers, shipping docks, stock rooms, and stairwells. Its mic picks up sound 30 feet away and plays them 1000 feet away. Combined with CCTV, it can record and play back a VCR.

Circle 17 on Reader Response Card

Integrated Speaker/Amp System

The PM 330C is the first in Numark's P/System 300 line. It incorporates a 15-inch woofer, a slotted port, and a 225W RMS amp within the same enclosure. Signals are routed to the unit via a 3-conductor balanced mic cable. For installed systems, the company notes that the PM 330C doesn't necessitate heavy-gauge speaker wires, amp racks, rack fans, and security covers.

Circle 18 on Reader Response Card

Multi-Application Mixer

Conneaut Audio Devices has made the Maxcon multiple application mixing system available. Available in rackmount and mainframe configurations, its uses include recording, sound reinforcement, post production, remote mobile, keyboard, theatre, and expansion console markers.

Circle 19 on Reader Response Card

Near Field Monitor

Tannoy's NFM-8, with Differential Material Technology, incorporates an 8-inch dual concentric driver in a ported, medite enclosure. Its suggested uses include small studios, broadcast facilities, remote recording vehicles, etc.

Circle 20 on Reader Response Card

30 Years Ago

Thirty years ago in *Sound & Communications*, we informed our readers of the benefits of background music in medical surroundings to increase patient confidence and decrease patient resistance to care and treatment. The article cited *The Journal of the American Medical Association* as saying: "...when it is used in the dentist;s office or in the hospital operating room, music reduces the amount of preanesthetic sedation and local or regional anesthetic required."

CCTV was "still dragging its feet" as a dealer item. Some reasons cited for CCTV's falling short of its predicted volume of $11 million were: its not yet being considered as a packaged item; high leasing prices; dealer/supplier competition; and applications not being as wide as sound system or packages. To combat the situation, CCTV was being worked over for "simplicity, smaller size, shortcuts in design and engineering." It was estimated that within a year newer systems would be about 30 percent cheaper.

Just introduced to the market was Bogen's BT400 high-power transistorized speech amplifier, said to project a message for a mile, with intelligibility that improved with distance. Made for use at airports, over water areas, on construction sites, in police and fire emergency work, it took two years of research to develop the unit. New from Audio-Lite was a loudspeaker baffle lighting fixture for houses of worship. It was available in styles such as Gothic, classic, modern, and colonial.

Meanwhile, the Communications & Sound Group, Electric Association had just announced its calendar for 1959-60. Meetings, which were open to "visiting firemen," were held at the Furniture Club of America in Chicago.

CALENDAR

Upcoming Events

OCTOBER

22nd Annual IICIT Connectors and Interconnection Technology Symposium: Philadelphia, PA. Contact: 312-940-8800. October 15-18.

Hong Kong International Electronics Fair: Hong Kong. Contact: 212-838-8688. October 16-19.

Georgia Tech Education Extension Course: Architectural Noise Control: Nashville, TN. Contact: 404-894-2547. October 17-18.

Syn-Aud-Con Sound Engineering Seminar: Secaucus, NJ. Contact: 812-995-8212. October 17-18.

87th AES Convention: New York, NY. Contact: 212-661-8528. October 18-21.

Georgia Tech Education Extension Courses: Audio Security: Boston, MA. Contact: 404-894-2547. October 19-20.

Society of Motion Picture and Television Engineers: Los Angeles, CA. Contact: 914-761-1100. October 21-25.

Georgia Tech Education Extension Course: Environmental Noise Control: Dallas, TX. Contact: 404-894-2547. October 26-27.

Security Show: Los Angeles, CA. Contact: 213-376-8878. October 25- 27.

Syn-Aud-Con Sound Engineering Seminar: Rockville, MD. Contact: 812-995-8212. October 26-27.

SPEECH INTELLIGIBILITY

(continued from page 65)

CONCLUSIONS

At the moment, the tools at hand to predict and analyze intelligibility are less than all one could hope for. Certainly the introduction of analyzers that can measure frequency response, RT 60, STI, RASTI is an important step forward in test instrumentation. But the near term future is even more promising with the trend towards the interface of sound system engineering software that "talks" to test instrumentation and provides aural output.

SOFTWARE UPDATE

Next month we will review two NCAD programs (non-Computer-Aided-Design). These are for quick estimating and also for those of us that have computerphobia—fear of computers. Altec's (Mark IV Audio) Acousta-CADD is scheduled for December and January. NEXOCAAD will be reviewed in February and March. Capsule reviews are in the works on speaker design programs, including Carvin's AudioCAD, CMS Leap, SDS CASD and CACD. A few of the additional programs being tested are flow charting programs, bid preparation software, and computer-aided-drafting.

Coming Soon . . .

Altec has postponed its release of Acousta-CADD 1.0 until after the AES Convention in order to put a some finishing touches on it. Acousta-CADD has been in development for years and will be an important new program. JBL has also postponed the release of CADP 4.5, the enhanced version of the program we reviewed last January. CADP II, JBL's next generation sound system design program is now scheduled for mid-1990.

NEXOCAAD, a program developed by IRCAM, the French acoustic and telecommunications research organization and the NEXO company, has also been delayed until after the AES Convention. NEXO decided to make some revisions on how the sound intensity was displayed on the floor plan. At the AES, they will announce a U.S. distributor for the program. NEXOCAAD has some exciting innovations, such as a high speed alternative to ray tracing for generation of the energy-time curve (which can be used to derive STI and RASTI intelligibility scores). Like the CASE program under development by Source Phoenix, NEXOCAAD is expensive and is intended for architectural acoustics engineering work in addition to sound system design. NEXOCAAD uses a Windows user interface, the same as JBL's CADP II, and runs on MS-DOS IBM compatibles. Bose should be shipping Rackmaker, its rack layout program, by the time you receive this issue.

BEARSVILLE

(continued from page 46)

HME-wired intercom system includes five dedicated permanent points — onstage, projection booth, downstairs prop room and one in each of the downstairs dressing rooms. In addition, the users can plug into the intercom in other locations near mic panels.

Rothstein also wired the room for five channels of Dolby Pro Logic sound for the 35mm projection system scheduled to be installed this winter (the original projection system, purchased in the mid-70s, was never set up and simply rusted away in storage). The screen will drop from the stage ceiling as will the flown sound reinforcement cluster (located directly behind the screen). This same flown cluster will be used as the primary sound source for the room PA, with sub-basses mounted in cavities on either side of the stage. Its design will permit the downward angle to change as it's lowered, keeping it properly aligned with the audience.

As of this writing, the full sound system has yet to be installed or even purchased (another $200,000 or so is required and at this time, no one knows where that money will come from). Rothstein's wiring allows the primary theatrical tenet, River Arts Theater Group, to use its own small sound system. Musical productions simply rent or bring in their own PA. So while the building now operates as a functioning theater, there's still more to come.

But he — what's the rush? If it's taken 15 years to get this far, what's a few more years going to matter? Called it a Woodstock state of mind — it'll be done when it's done. Asked to estimate to possible project completion date, sound contractor Rothstein smiles and scratches his chin.

"Ohhh...maybe another 15 years. I expect to be working on this again in the year 2000."

AD INDEX

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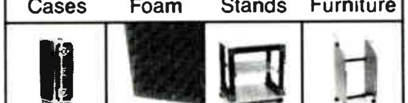

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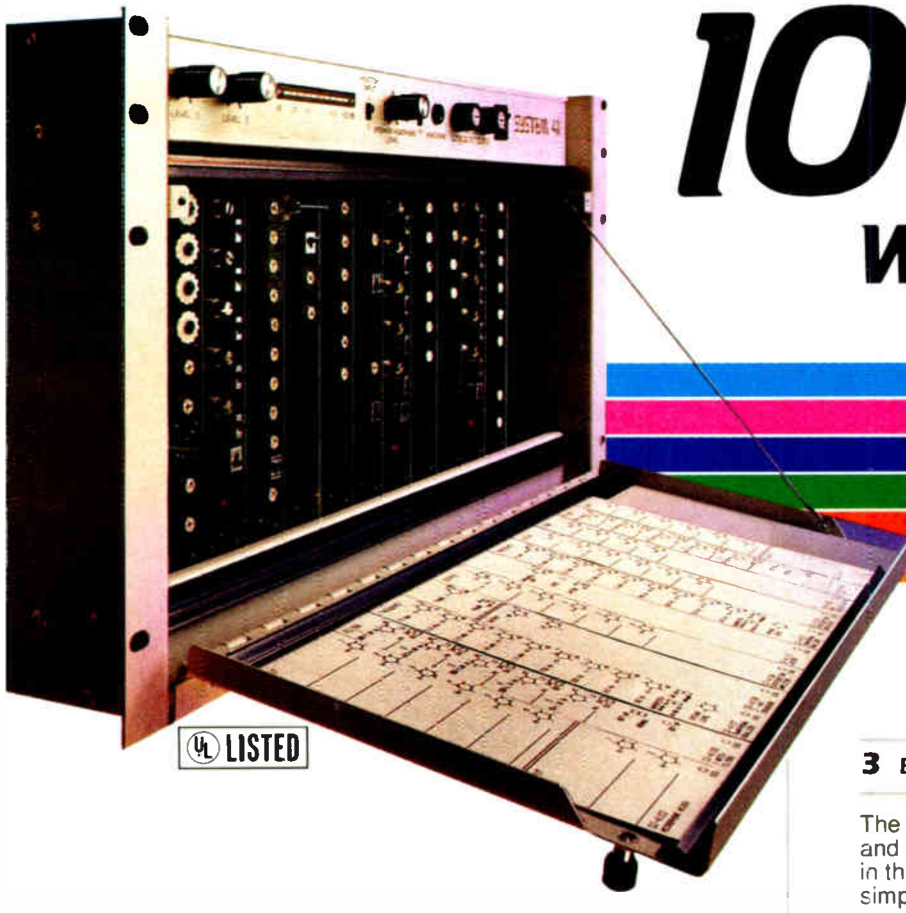

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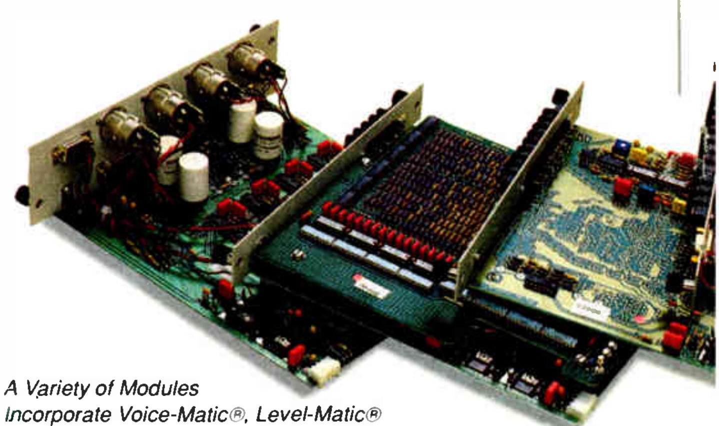

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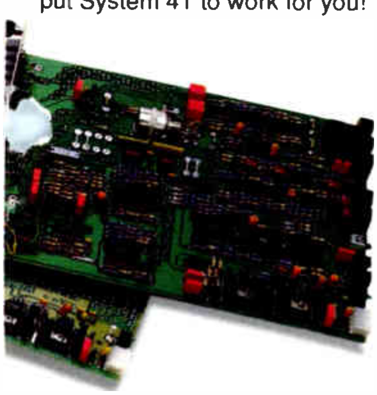
8 DEDICATED CUSTOMER SERVICE

We know that once you reach a "comfort level" with System 41, it will be your system of choice. That's why IRP sales and technical staff are just a free call away: **1-800-255-6993**. Or FAX us your system requirements. You'll get a prompt reply on how to put System 41 to work for you!



9 PATENTED TECHNOLOGY

System 41 incorporates state-of-the-art technology that has become industry standards. Available as modules are the Voice-Matic® Automatic Microphone Mixer; Level-Matic® Automatic Level Controller, TEQ® Transversal Equalizer and the Audio Signal Delay.

10 IRP . . . A PROVEN LEADER

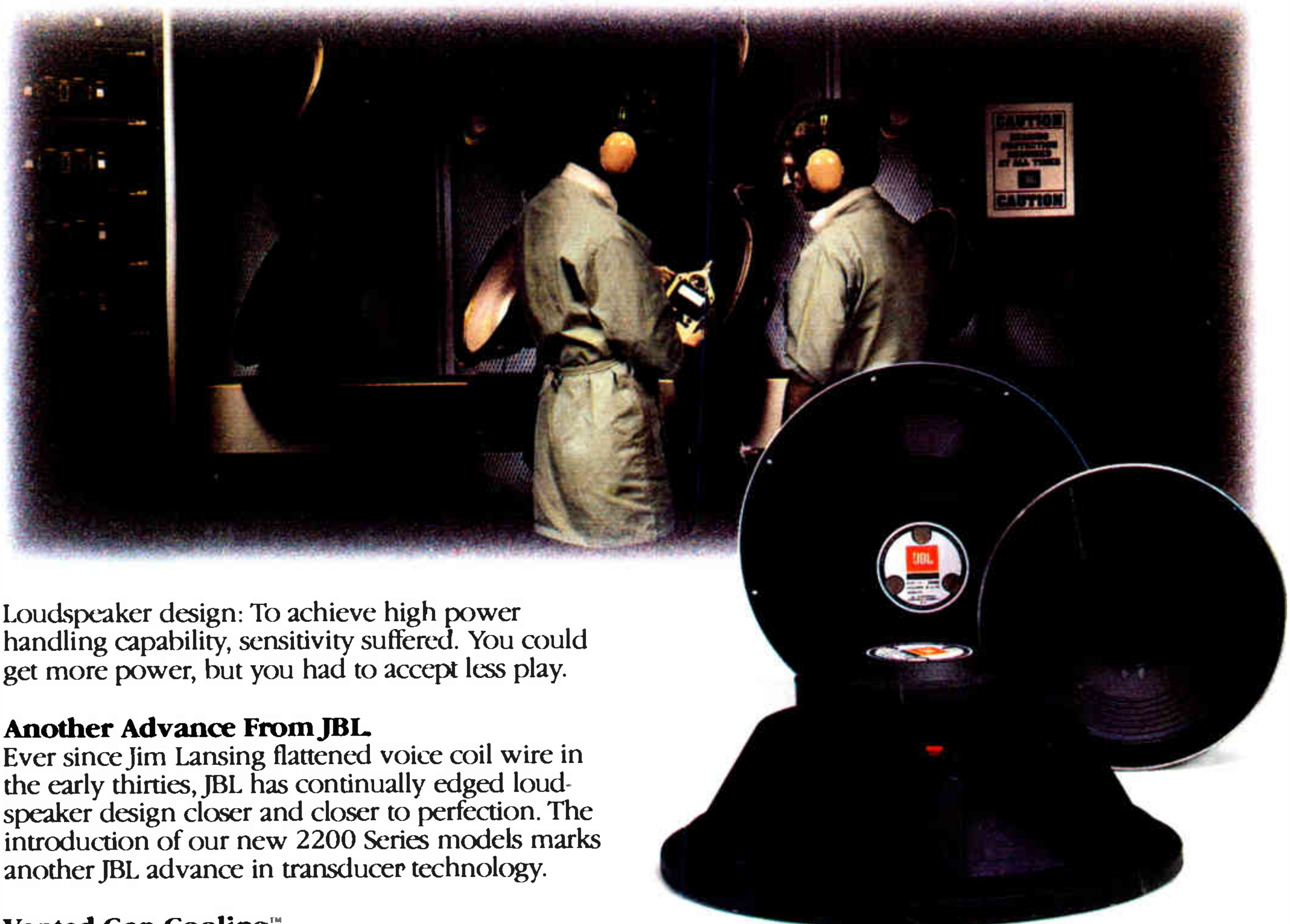
Leading sound contractors and acoustical consultants have specified IRP products for their innovative design, quality performance and reliability. System 41 inherits that tradition of excellence.

IRP industrial research products, inc.
A Knowles® COMPANY

415 Busse Road, Elk Grove Village, IL 60007
Phone: 312-640-9553 or 1-800-255-6993
FAX: 312-640-9607

Circle 202 on Reader Response Card

Power. Play.



Loudspeaker design: To achieve high power handling capability, sensitivity suffered. You could get more power, but you had to accept less play.

Another Advance From JBL.
Ever since Jim Lansing flattened voice coil wire in the early thirties, JBL has continually edged loudspeaker design closer and closer to perfection. The introduction of our new 2200 Series models marks another JBL advance in transducer technology.

Vented Gap Cooling.™
As a voice coil's temperature rises, so does its impedance, causing power compression. Every dB of power compression translates into one less dB of output.

To reduce power compression and gain overall sensitivity, JBL created vents from the voice coil, through the magnet structure, to the rear of the loudspeaker. As the cone travels, heat is dispersed and cool air returned to the voice coil. The result, a true 600 watt loudspeaker with high sensitivity. You get twice the power handling plus greater output.

New SFG™ Magnet Structure.
Through computer modeling and simulation, the 2200 Series SFG magnet structure achieves optimum flux density and significantly lower distortion while realizing up to 20% weight reduction.

We Heard You Loud And Clear.
Whether you're looking for a mega-watt touring loudspeaker, or for a low distortion critical listening device, JBL has the model for you. All with the new SFG magnet structure with Vented Gap Cooling. The new 2200 Series models give you the best of both worlds. Power, and play.

JBL

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