NSCA Contractor's Expo '86 in Review

SOUND & COMMUNICATIONS

FOR CONTRACTORS, SYSTEM MANAGERS AND SPECIFIERS

JUNE 1986

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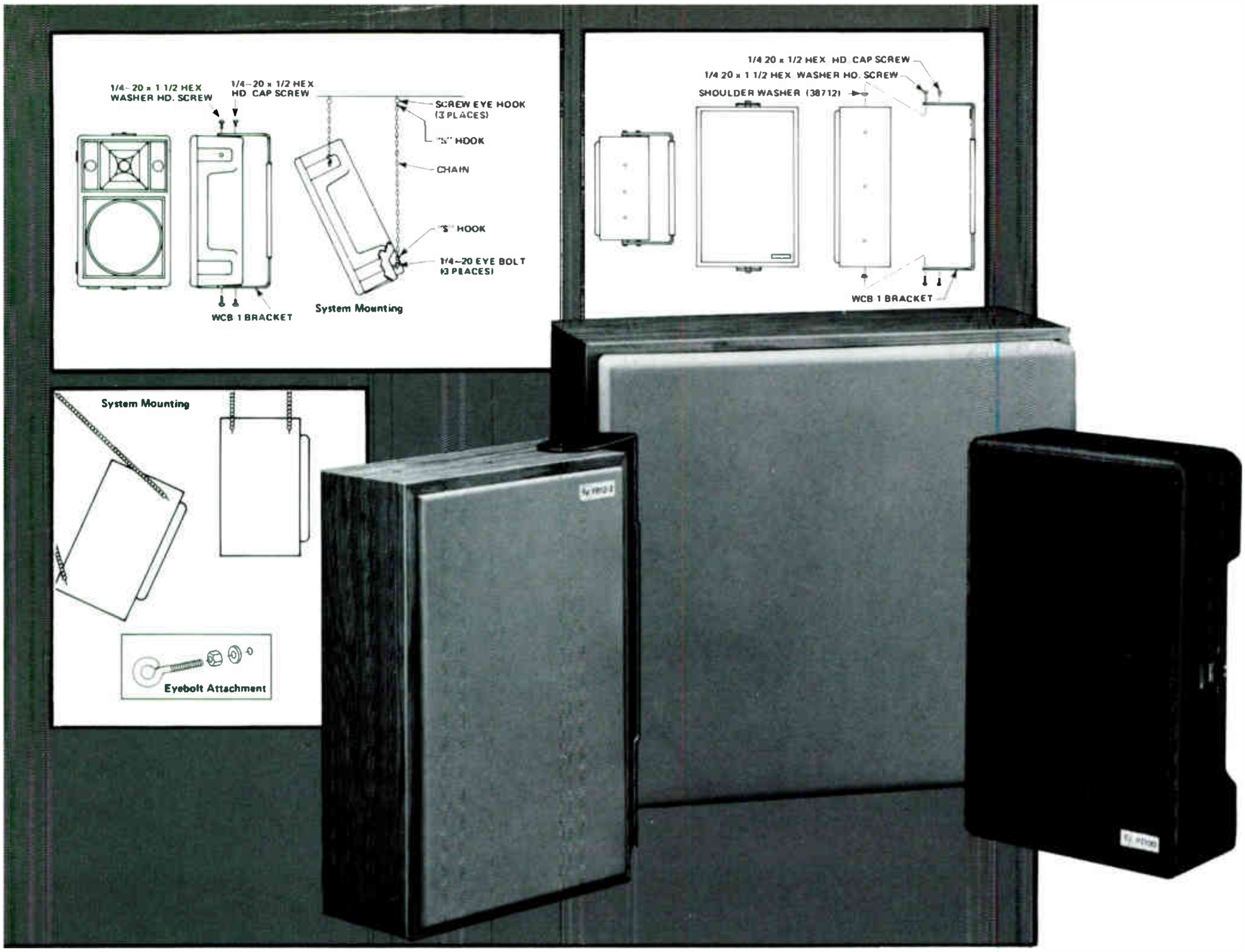

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All systems are factory-fitted with threaded inserts to facilitate suspension. And, with an optional telescoping bracket, the FR12-2 and PI100 can also be wall or ceiling mounted in six versatile positions. For constant-voltage operation, an optional TK60 line transformer kit replaces the normal direct input panel.


TK60 TRANSFORMER KIT

The FR15-2 and FR12-2 have oak-grained, vinyl-covered enclosures, for use indoors.

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The FR15-2, FR12-2 and PI100 speaker systems from Electro-Voice. Outstanding performers that install with ease and look as great as they listen. Let us tell you more. Contact Jim Long, Director of Marketing/Professional Sound Reinforcement, Electro-Voice, Inc., 600 Cecil St., Buchanan, Michigan 49107.

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SOUND & COMMUNICATIONS

Volume 32 #6

June 1986

16

FEATURES

COLUMNS

DEPARTMENTS

ON THE COVER

The June issue of *Sound & Communications* examines different ways of measuring and testing sound. On the cover is some of the equipment from the audio workstation at Andrews Audio Consultants, a New York City-based sound contractor. Shown is equipment from Techron, Hewlett-Packard, and an Apple–based FFT system.

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IDEAS & VIEWPOINTS

Technically Speaking

Editor's Note: In the beginning of this year, we at Sound & Communications *took a look at the previous year's successes and failures. We felt good at what we had accomplished in the first year of editing and publishing* Sound & Communications, *but we realized there was even more we could do. We began looking for new and better ways to serve you, our reader. We found ourselves speculating about the contracting industry, as were many of our peers, because there was no hard data available to confirm our ideas or to correct them. So, we decided to acquire our own hard data and do a market survey. That survey, done in conjunction with NSCA, appeared in last month's* Sound & Communications*!*

In looking at the past year, we had also felt that Sound & Communications *could improve the technical information we thought a magazine of this caliber should have. So, we looked around for a Technical Editor and found one right underneath our noses. We hired Jesse Klapholz, one of our own freelance writers and an acoustical consultant. But we also knew that no one person could be an "encyclopedia of sound and communication knowledge." So, based on Jesse's recommendation (and hard work), we put together a Technical Council of experts from various areas of the industry. The Council will be called upon to play an active role in the development of* Sound & Communications. *They will be asked to contribute ideas as well as articles and, on occasion, review articles written by others for accuracy and content.*

During the recent NSCA Contractor's Expo, we wrote, produced and edited NSCA-TV, a daily news and information show for all attendees of the NSCA Contractor's Expo in Las Vegas. On-the-air 24 hours a day, NSCA-TV became everyone's daily reminder of what to see and do.

So far, 1986 has been a year of goals and accomplishments for us at Sound & Communications. *With your continued support, we will succeed in achieving all our aspirations for the success of all of us.*—NP

Getting Bigger and Better

When somebody gets involved with a new job it is quite easy to come up with a long list of dream-filled ideas—almost like a Channukah (or Christmas) shopping list. One of those ideas was to form a *Sound & Communications* Technical Council. The Technical Council, a group of sound and communications professionals from academia, consulting, and contracting, would serve as an active forum for the contribution and exchange of information. Starting this month *Sound & Communications* is proud to announce its new Technical Council (of which, two members, Jaffe Acoustics and Bill Thornton, have contributed to this issue).

While we are all so busy running around selling, bidding, designing, and installing, we at *Sound & Communications* thought it would be educational to read the story from the "other side of the tracks." Therefore, you will also notice, beginning with this month's *Sound & Communications*, a column—Consultant's Comments written by Marc Beningson of Jaffe Acoustics.

The challenges of the 1980s, with the ever increasing merger of technologies, industries, and markets, are certainly having a great impact in the sound and communications industry. In the uncertainty and unclarity of our market one thing is clear—change. With an industry changing as rapidly as ours, it is most important to be sensitive to the overall direction of our past, present, and future.

As a journalist, a "hands on" operator, and designer of audio systems, I look forward to the challenges afforded by this new position. This is *your* magazine, and as such what you have to say is most important to all of us in the business. As Technical Editor of *Sound & Communications*, I want to hear your ideas.

With the services *Sound & Communications* already supplies, an annual BLUE BOOK, an economic survey, and the marketing report, plus more informative features and columns, and the formation of a new Technical Council, *Sound & Communications* reinforces its commitment to the industry. I wholeheartedly accept the opportunity to be a member of the *Sound & Communications* staff, the original sound and communications magazine, bringing together, through this "forum in print," the specifier, manufacturer, contractor, and systems operator.—JK

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.

Take us apart for a side-by-side comparison of ceiling baffles, for example. From the heavier gauge metal to the more durable epoxy finish, Quam baffles are made in our own plant to look, install and perform better. The same is true of the entire Quam line, from enclosures to transformers to 8" speakers.

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HANDS ACROSS AMERICA USES SOURCE PRIVATE NETWORK FOR COMMUNICATIONS

Source Telecomputing Corporation provided Hands Across America with a private network for nationwide computer-based communications. The private network, called HAANET, facilitated the exchange of fast-changing and time-critical information among Hands Across America's geographically dispersed staff. Accessible with any standard personal computer, modem, and telephone, HAANET enabled staff to retrieve and send messages instantly, share information to solve logistical problems, and get information on nationwide weather, late-breaking news, and airline schedules.

RESEARCH FIRM PREDICTS RAPID SALES GROWTH FOR T-1 PBX PERIPHERALS

Sales of PBXs with T-1 PBX-to-host-computer interfaces will increase approximately 200 percent over the next 10 years, according to a study recently published by International Research Development, a market research firm. According to Leslie Townsend of International Resource Department, while PBX sales, due to the large base of installed PBXs, will not increase over the next few years, the peripherals market provides the PBX vendor with high sales potential. He said peripherals can be sold with higher margins than PBXs because the vendor has a captive market. But Townsend added that in the long run revenues from PBX peripherals sold as add-ons will be less significant than the growing sales of PBX units with peripherals.

BOTH BUSINESS WEEK AND INC. INCLUDE V BAND AMONG HOT SMALL PUBLIC CORPS.

V Band Systems Inc. of Yonkers, NY, has been getting a lot of attention lately from the business press in particular. Business Week recently listed it as the tenth in its Top 100 best small corporations. Also, Inc. magazine listed V Band as 22, in its 1986 Fastest-Growing 100 Small Public Companies. According to Business Week, the manufacturer of electronic key phones for trading floors achieved $21.2 million in sales over its most recent 12 month report period. It's three-year average showed a 116.6 percent increase in sales with a 218.6 percent increase in profits. Inc. put V Band's sales growth from 1981 to 1985 at 5,648 percent, a compounded annual rate of 175 percent.

KLEIN TOOLS, INC. ACQUIRES VACO PRODUCTS CO.

Klein Tools, Inc., a Chicago-based producer of hand tools and occupational test equipment, has acquired Vaco Products, Co., also of Chicago, a manufacturer of screwdrivers, nut drivers, wire terminals, wire connectors, and other hand tools. A joint announcement by Mathias A. Klein, chairman of Klein Tools, and Ramond Sailverstein, president of Vaco Products, stated, "No changes are contemplated in the management, organization, or marketing programs of either company and both companies will continue to sell under their independent brand names and through their individual sales and marketing groups."

AIPHONE U.S. MANUFACTURING PLANT BEGINS PRODUCTION

Aiphone Corporation has begun assembling selected intercom systems in its new Bellevue headquarters facility. The assembly capability is being added to help the company meet the demands of its expanding U.S. business, according to Hiko Shinoda, president of Aiphone Corp.

The first intercom product to be assembled at the Bellevue plant is the GX-300 Drive-Thru, an intercom system for fast food restaurants, banks, photo processing stores, gas stations, etc. Aiphone Corp. decided to produce the GX-300 in the Bellevue facility because the market for the product is in the United States and Canada. "It's a North American product that meets a North American need," said Shinoda. "There aren't anywhere near the drive-through businesses in other countries."
Systems parts will also be North American, with 98 percent of components for the

GX-300 acquired from domestic sources, Shinoda said. The GX-300 will not be the only product assembled at the Bellevue plant. Special order and custom intercom systems are already assembled there, Shinoda said. The company said it plans later to add to the number of intercom systems produced domestically for the American Market.

DEMCHUK OF TELECO USA TO PROMOTE SALES OF CYBER DIGITAL PBXS

Under a joint agreement between Teleco USA, Inc. and Cyber Digital, John Demchuk, national director of network development for Teleco USA, will act as a sales and marketing consultant for Cyber Digital to promote sales of the Cyber Digital Data/Voice PBX to the Teleco USA network. Demchuk will also counsel the Cyber Digital marketing department on sales of the Cyber Digital MSX PBX to other vendors. Cary Masi, chairman of the board and chief executive officer of Teleco USA, said, "This is a giant step forward for Teleco USA in intensifying its national marketing programs. John Demchuk has over 25 years of telecommunications marketing experience and is a valuable member of our marketing team."

TOSHIBA SIGNS $80 MILLION DOLLAR AGREEMENT WITH USX TELECENTERS

The Telecommunication Systems Division of Toshiba America, Inc. Has signed a marketing agreement with USX Telecenters, the Sunnyvale-based chain of franchised business telephone centers. Under the terms of the $80 million, multi-year agreement, Toshiba will provide USX Telecenters with Toshiba telephone systems and equipment, according to Paul Wexler, TSD's vice president and general manager. The agreement calls for Toshiba to supply privately labeled electronic key and PBX systems to the newly formed telecommunications company. The two firms also agreed to joint development of proprietary features and enhancements.

WILLIAMS SOUND CORP. MOVES TO LARGER FACILITY IN MINNETONKA, MN

Williams Sound Corp., manufacturer of wireless hearing assistance systems, wireless microphones, and tour guide systems, has moved from its Eden Prarie, MN, location to a larger facility (at 5929 Baker Road, Minnetonka, MN, 55345-5997). The new facility, which is twice as large as Williams' previous building, is part of the Baker Technology Plaza.

NEW SOUNDOLIER ENCLOSURE FACILITY TO SERVE EAST COAST DEMAND

Soundolier, A Division of American Trading and Production Corporation, has opened a new manufacturing facility in Laurinburg, NC, to build enclosure systems for distribution on the East Coast. The multimillion-dollar facility is located on a 13-acre site, and provides 50,000 square feet of manufacturing area, according to the company. It has been equipped with latest technology for producing metal consoles and racks, including a CNC turret process, punch presses and MIG-welding equipment, shearing and press brakes, as well as a conveyorized paint system. The facility supplements Soundolier's prime manufacturing site in Festus, MO. A spokesman said the plan was added to meet rising demand in the East and to reduce shipping costs to customers there.

MORE THAN 200 NEW PRODUCTS AT USTSA EASTERN TELECOM CONFERENCE

More than 200 new products were introduced at the 1986 Eastern Telecommunications Showcase, May 20 to 22, at the Georgia World Congress Center in Atlanta, GA, according to Donald R. Pollock, managing director of the United States Telecommunications Suppliers Association and manager of the show. The new products included automatic meter reading equipment, fiber optic vans, test equipment, and turnkey systems. To facilitate finding the new products, each attendee received a list showing companies, products, and booth numbers.

by S. Ann Earon
Telemanagement Resources Int'l.

ASSESSING YOUR CLIENT'S NEEDS

Too often vendors of products and services position themselves to potential customers as offering the best product or service to meet the customer's needs. But how does a vendor assess a potential client's needs in a manner that convinces the client to use the vendor's products and services? Conducting a "needs assessment" is one way for vendors to convince clients of the value of a specific product or service.

A needs assessment is a methodology designed to analyze the way an organization conducts business. The purpose of a needs assessment is to determine what specific business needs exist within an organization and how a product or service will meet those needs. By positioning products and services to meet specific needs, vendors set themselves apart from their competitors. Potential clients understand the value of meeting their needs.

"The purpose of a needs assessment is to determine what specific business needs exist within an organization and how a product or service will meet those needs. By positioning products and services to meet specific needs, vendors set themselves apart from their competitors."

This article is designed to teach vendors how to assess client needs. Although the needs assessment process applies to many situations, in this instance it will be related to the field of teleconferencing.

Teleconferencing, defined as two or more people communicating electronically from locations separated by distance or distance and time, is a cost-effective alternative to many face-to-face meetings. With advances in technology, more and more vendors are entering the marketplace with teleconferencing products and services. Potential users of these products and services are often confused by the claims made by vendors. To win customers, and more importantly, repeat business, vendors must begin to approach clients differently. A needs assessment offers that change in approach.

Benefits

In order to design and implement a successful teleconferencing system, it is important that a thorough analysis of an organization's needs be undertaken. A needs assessment provides value to a client in four ways: (1) It provides the data needed to develop economic justification for determining costs and benefits. (2) It provides input into the system design phase by identifying functions, locations, and specific equipment needed. (3) It helps to assure that once a system is installed it will be used effectively. (4) It provides input into long-range plans for system expansion.

In other words, a needs assessment provides the client with ammunition needed to justify the dollar expenditure you, as the vendor, are requesting the client to make.

While a needs assessment takes time to complete, the benefits make the effort worthwhile. Too often people think they know what they need and myopically install equipment to meet preconceived notions. Two problems arise: (1) the equipment installed does not meet the needs of the client, or (2) the equipment installed only meets some of the needs, while overlooking others. The benefits are greater to the customer and to the vendor if all the needs are met.

Methodology

To conduct a teleconferencing needs assessment a questionnaire is designed to assess a client's meeting and travel patterns. By analyzing existing patterns it is easier to develop a recommendation to meet identified needs. For example, if a client frequently has employees attending meetings where objects or drawings are viewed, it is important to include graphic support systems as part of the recommendation. If, on the other hand, individuals are frequently traveling between locations to see and talk with other individuals, without the need to look at charts or objects, then visual eye-to-eye communication is important. Too often vendors push their product or service without considering the client's needs. Often an organization that installs a teleconferencing system without assessing the need for it, will find the system underutilized.

To effectively assess an organization's needs it is important to interview a cross section of people within the organization. Individuals in engineering and manufacturing, management and administration, sales and marketing, and training functions are typically potential users of teleconferencing. This does not mean that people in other job functions, like finance and personnel, won't use teleconferencing, but they are not usually the drivers of the usage.

To conduct a thorough needs assessment, 15 to 20 percent of the target population should be interviewed—either face-to-face or with a mail-in questionnaire. As individuals are interviewed, travel data should be collected. The data sought is related to the number of trips between frequently traveled locations and expenses associated with those trips. Once all the data is collected an economic cost justification must be made.

Economic Justification

While many of the true benefits of teleconferencing are difficult to quantify—increased productivity, better use of a manager's time, sharing of scarce talent—it is important to the client to have the expense of a teleconferencing system justified.

Teleconferencing can easily be viewed as a supplement to travel. However, the displacement of travel (whether real or imagined) is one factor that is important to controllers of most corporations.

An economic justification can be designed to look at several teleconferencing factors. Three factors typically accepted by one or more decision makers within a corporation are (1) travel displacement, (2) increased productive time, and (3) improved efficiency of a manager's time.

Travel Displacement: While it is not realistic to assume that teleconferencing will displace all travel, it is reasonable to assume that teleconferencing may displace some travel. The Travel Cost Displacement model looks at what percentage of existing travel can be displaced through the introduction of teleconferencing.

Value of Lost Time: In addition to looking at Travel Cost Displacement, it is important to look at the amount of time lost while traveling. This data is available by tallying and averaging responses received in the questionnaire.

In addition to assessing travel and time dollars, it is important

(continued on page 52)

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☐ G Advertising/Promotion
☐ H Other ______________

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☐ 1 Contractor—Engineered Sound/Acoustical
☐ 2 Contractor—Interconnect/Intercom
☐ 3 Contractor—Fire/Alarm/Safety
☐ 4 Electrical Contractor
☐ 5 Pro Audio/Studio/Reinforcement
☐ 6 Architect/Designer
☐ 7 Engineering/Acoustical Consulting
☐ 8 Maintenance/Service
☐ 9 Dealer/Distributor/Rep
☐ M Manufacturer
☐ 0 Other ______________

④ **Your purchasing authority is:**
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☐ Recommend/Specifier
☐ No Direct Authority/User

⑤ **Intensity of your product need:**
☐ Have salesman call
☐ Need within 3-6 months
☐ Future projects

⑥ **Number of employees at your company:**
☐ 1-3 ☐ 4-10 ☐ 11-25
☐ 26-100 ☐ over 100

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☐ E Field Installation/Maintenance
☐ F Consultant
☐ G Advertising/Promotion
☐ H Other ______________

③ **Primary business of company (only one):**
☐ 1 Contractor—Engineered Sound/Acoustical
☐ 2 Contractor—Interconnect/Intercom
☐ 3 Contractor—Fire/Alarm/Safety
☐ 4 Electrical Contractor
☐ 5 Pro Audio/Studio/Reinforcement
☐ 6 Architect/Designer
☐ 7 Engineering/Acoustical Consulting
☐ 8 Maintenance/Service
☐ 9 Dealer/Distributor/Rep
☐ M Manufacturer
☐ 0 Other ______________

④ **Your purchasing authority is:**
☐ Final approval/Buyer
☐ Recommend/Specifier
☐ No Direct Authority/User

⑤ **Intensity of your product need:**
☐ Have salesman call
☐ Need within 3-6 months
☐ Future projects

⑥ **Number of employees at your company:**
☐ 1-3 ☐ 4-10 ☐ 11-25
☐ 26-100 ☐ over 100

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by Marc Beningson

WORKING WITH SPECIFIERS

In the building industry, a specification is used to represent the portion of a facility's design to a contractor who becomes responsible for the execution of that design. To maintain consistency among trades, the specification format and the administration process have become highly standardized. These standards have been developed by the construction industry—architects and contractors—over a period of many years and are appropriate for the basic elements of construction such as masonry, steelwork, studwalls, electrical, and mechanical installation work. The process of bidding, award, and administration of contracts, coordination of work among trades, and acceptance of completed work is reasonably similar for these aspects of construction and, more importantly, the implications of different designs and installation techniques are well understood throughout the building industry.

Over the past five to 10 years, a number of new aspects of construction have been developed. Many of these are high-tech or of a specialty nature such that the standards must be expanded or modified to accommodate these new aspects. These include computer networks, intelligent building control systems, and complex communication systems. Although sound systems are not new to facilities such as theaters, performing arts centers, nightclubs and corporate boardrooms, sound systems have reached a level of technology and sophistication such that modifications to the traditional process of specification and contracting are required to provide the owner with a successful installation.

Incorporating these changes into the basic construction process requires the cooperation of architect, project engineers (primarily the electrical engineer), and the sound system design consultant. Based on continued involvement with this process, the following are concepts of some minor adjustments which may be easily incorporated into the process and result in major improvements in the quality of the completed project.

(1) Eliminate the performance specification. Twenty years ago the level of technology was such that it was adequate to specify a level of performance, but the complexity required by most users today demands that a design be completely specified. This does not mean that specific equipment must be listed—acceptable equivalents or alternates may be included—but at very least, a block diagram should demonstrate the operational design intent of the system.

(2) Pre-bid qualification of contractors can ensure that only specialized sound system installers with experience in projects of similar scope or magnitude may bid. Thus, unqualified contractors cannout influence bidding with an inappropriate price that does not truly represent the workscope.

(3) A pre-bid conference with all qualified contractors and the system designer should be held by the bidding authority (owner, general contractor or construction manager) about two-thirds of the way through bidding. This permits the contractors to directly question the designer instead of submitting requests for clarification to administrators who do not necessarily understand the specifications. Additionally, the designer may issue clarifications or even corrections to the specification documents in a manner that ensures that all contractors receive the same information. In this way, all contractors submit bids based on an identical workscope rather than interpretation of design intent.

(4) After acceptance of qualified contractors to the process, **open lines of communication should exist between contractors and the design consultant.** All phone conversations should be documented and published to all involved bidders at least 10 days prior to the close of bidding, again to ensure that all contractors submit bids based on an identical workscope. When this is so, it is much more certain that the lowest bid will be an acceptable bid. After award of the sound system contract, there must be continued communication between contractor and design consultant. Although all decisions must be documented through the process of submittals to the architect and general contractor or construction manager, the process itself should not preclude a close contractor/design consultant relationship. Cost savings, design improvements and substitution of new products may result from such dialogue, all of which are in the owner's best interests.

(5) The contractor must submit concise and detailed shop drawings to demonstrate that the design intent is understood and to indicate clearly the intended details of execution. The drawings should include a three-wire schematic (of a sample circuit in the case of large or complex systems) and should not include direct copies of the specification documents. Shop drawings should show, to the highest level of detail, wiring and grounding techniques, hardware, mounting means, dimensions, and locations of devices relative to known points as accurately as possible.

The design consultant must review these drawings carefully for conformance with the design intent. Many contractors seem to shy away from detailed shop drawings because of the time (and therefore cost) involved. However, the purpose of shop drawings is for the contractor to demonstrate exactly what is to be provided. As such, when the drawings are approved, the contractor has received authorization to proceed with a specific methodology. Of course, if the execution of the system adheres closely to the shop drawings, the production of as-built drawings—which should be an integral part of every specification—will require little effort.

(6) In large systems, the specifications—and the design consultant's contract—should include a provision for an inspection of the complete rack assemblies, which, of course, should be fully assembled and tested at the contractor's shop prior to field installation. Here the design consultant and contractor can perform basic operational tests and identify problems of hum, RFI, and grounding, while the racks are isolated from the building's electrical system. Once the racks are installed on site, these tests will be repeated, but more basic problems such as miswired circuits and defective equipment are more easily resolved while the racks are still in the contractor's shop.

(7) There must be a means for the design consultant and sound contractor to **ensure that work performed by other trades—primarily electrical, but also mechanical and structural steel contractors—meet the requirements of the sound system**. Largely, these requirements should be set by the design consultant in conjunction with the other project engineers and consultants during the early stages of the project, but the sound contractor should bring any observed potential problems to the attention of the general contractor or construction manager.

(8) The sound contractor and design consultant must be provided with sufficient time to test and tune the sound system prior to first use. During the final weeks of an extensive construction

(continued on page 51)

by Michael Klasco

THE COMPUTER AUDIO WORKSTATION

Traditionally, the sound contractor has relied on "dedicated" test equipment for acoustic measurement. The term dedicated implies that the equipment is limited to a specific function, such as sound pressure level meter, volt meter, oscillocsope, or spectrum analyzer. An alternative to dedicated test equipment is the personal computer-based audio workstation.

Application software programs and hardware that turn PCs into PC audio workstations, has been developed from numerous sources that expand the capabilities and productivity for the sound contractor. Software from the engineering sector which can be used in the PC audio workstations are categorized as Computer-Aided-Engineering (CAE), Computer-Aided-Drafting (CAD), Computer-Aided-Test (CAT), and Computer-Aided-Manufacturing (CAM).

Software

Sound system designers can use CAE for cluster layout, RT60 predictions, bass enclosure alignment simulations, and crossover network design. CAD is used for drafting functions such as wiring diagrams, equipment rack layouts, and cluster drawings. CAT can be used for one-third-octave spectrum analysis, FFT analysis, RT60 measurements, multimeter, distortion, and oscilloscope functions.

An appealing feature of the PC audio workstation is its building block capability. With plug-in boards, distortion analysis or oscilloscope functions can be added to an existing system. Color and speed of printouts can be enhanced with an X/Y plotter. And drafting can be made less tedious and more accurate with a mouse or graphics tablet.

Computer-Aided Drafting

The power and speed of computer drafting is derived from two aspects: the building of drawings from pre-drawn "macros," and the ability to revise or edit previous drawings into new drawings. For most sound system drafting, the basic components are often used on jobs. Specific mixers, amplifiers, equalizers, horns, bass boxes, etc. will be used in wiring diagrams, rack drawings, cluster designing, low frequency enclosure prints, and so on.

Most CAD programs use a template technique known also as "macro" or "block." Computer file libraries are initially created with all the commonly used components, each item with its own file number. Once the template file library has been created, high-speed drafting is performed by the computer drawing the component on the screen and the operator positioning it. If, for example, six amplifiers are needed in a rack, the operator requests the amplifier template and positions and locks it to the desired locations. Using templates, complete drawings can be created in a small fraction of the time it now takes with conventional drafting techniques. Complete system drawings are also stored in file libraries (on floppy or hard disk). When a new system is similar to a previously filed system, the operator accesses the file, edits and revises the file (similar to word processing) to create the new drawing. The original file remains intact and the new drawing is saved under its own name (or number).

Printouts of the file can be on a dot matrix printer or an X/Y plotter. Low budget systems can use 8.5 x 11-inch printouts spliced together and blueprinted until a large sheet, high performance X/Y plotter is acquired.

CAE is also described as computer simulations and computer modeling. The concept is to pretest prototypes (of speaker enclosures, cluster designs, architectural acoustics, circuit designs, etc.) before they exist or are modified, through computer simulations. "What if" variations are tried. (What if the enclosure volume is increased? What if the 90 x 60-degree horn was replaced by a 60 x 40-degree horn? What if 300 square feet of three-inch thick fiberglass was glued to the ceiling?).

Aside from the benefits of increased productivity and higher quality engineering, computer simulation programs are effective and powerful sales tools when used as part of your sales proposals.

Architectural Acoustics

Typically, CAE programs require room dimensions and respond with surface area and internal volume. Room modes are calculated and RT60 predicted. The more comprehensive programs contain expandable file libraries for materials and their absorption coefficients. With some basic guidelines, these programs work as an early warning system if the simulations fail outside of a predetermined tolerance. Acoustics II from Headware and Studio II are examples of these programs.

Speaker System Design

These programs aid in selection of low-frequency drivers and box tunings crossover network design and related "What if" comparisons trade-offs. Computer-aided-speaker-design (CASD) is available from Scientific Design Software.

Cluster Design

Of all the programs of use to sound contractors, sound system design has received the greatest visibility.

Programs fall into two categories, those that help engineers lay out the job and those that others focus on "testing" a prototype design. Because of the detailed coverage of cluster design programs in pro-sound trade and technical journals, I will limit atention to this topic. Software vendors or cluster design programs are JBL (CADP), E-V (Vamp), Bose, Menlo Scientific (ON-LP) and the Cluster Computer/Program by John Prohs.

Circuit Design Modeling

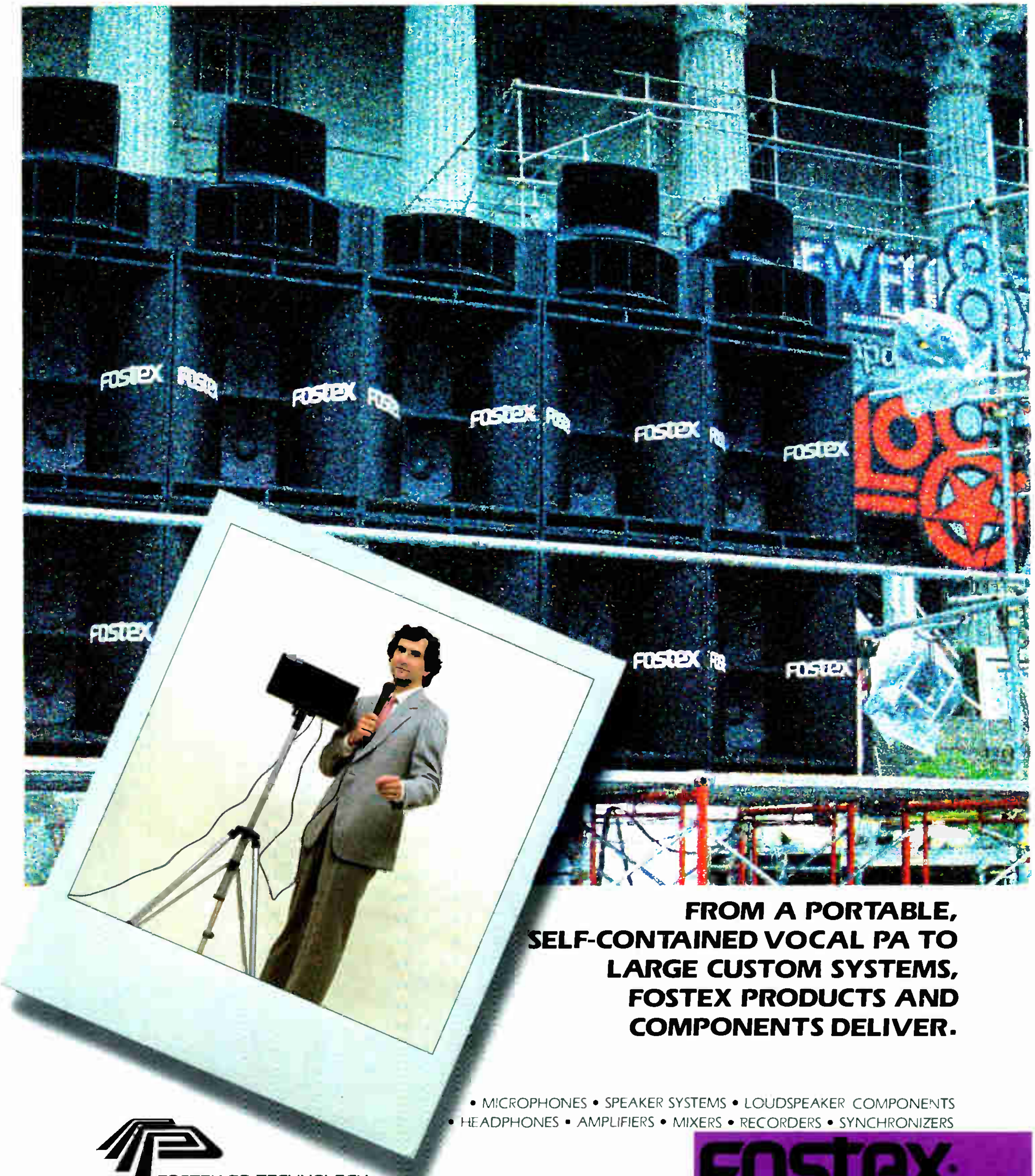
Sound system designers can use these programs to determine values for electronic crossovers, subsonic filters, and more ambitious projects. Programs vary, offering some or all of these features—schematic drawing, circuit test from sonematic, routing for circuit board layout, artwork for circuit boards.

Some vendors of circuit simulation software are Micro Cap and VD Engineerings. Circuit board layout software is available from Winteck and Orchard.

Computer-Aided-Test

Computer-based test equipment for audio has flourished in the last few years. Outboard boxes and plug-in boards for one-third-octave, FFT high-resolution spectrum analysis, reverberation time, harmonic distortion, intermodulation distortion, multimeters and more, are available from many suppliers.

Mike Klasco, president and founder of Menlo Scientific, has recently been working on the sound system for the 1988 Olympics in Seoul, Korea.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by William Thornton
Thornton Noise and Acoustics

SOUND INTENSITY AND POWER

Sound *pressure* level, sound *intensity* level, and sound *power* level are three distinct quantities. In the past, only sound pressure level was measurable. With recent advances in microprocessor technology, sound intensity may also be measured and sound power can be computed from it. This article explains the differences among these quantities and introduces the new technology of the "Sound Intensity Method."

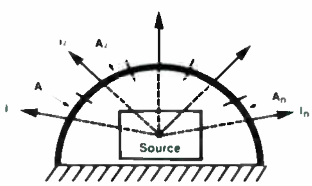


Figure 1: Sound Power

Work and Power

A review of physics will reveal the subtle differences among these quantities and show how they are related. Work is defined as force times distance. In acoustics, pressure times area is force, and work is expressed as:

Equation 1

Work = [P (t) x A] x dr

where:

P (t) = pressure,
A = area,
dr = distance.

Power is defined as the rate of work per unit of time:

Equation 2

Power = d (Work) / dt = [P (t) x A] x dr / dt

Velocity or V (t) is dr / dt, and power can be expressed as:

Equation 3

Power = [P (t) x A] x V (t)

Intensity is defined as P (t) x V (t), and equation 3 can be rewritten as:

Equation 4

Power = P (t) x V (t) x A = I (t) x A

where:

I (t) = intensity

Pressure, Intensity, Power

Sound pressure level is the time averaged pressure magnitude of a sound wave expressed in decibels. Its magnitude varies with distance. It is a scalar quantity, i.e. it has magnitude but no direction.

Sound intensity level is the time averaged power per unit area emitted by a sound source. The magnitude of intensity is expressed in terms of decibels and it is a vector quantity, i.e. it has magnitude and direction. It measures the flow of power per unit area in a specific direction.

Sound power level is the integral (summation) of the intensity times area for the entire surface area surrounding a sound source. It is the *total* amount of energy radiated from the sound source per unit of time. It does *not* vary with distance; it is a fixed quantity for the source. Referring to **Figure 1**, the total sound power can be computed by multiplying intensity times area around the entire surface area. In equation form, it is expressed as:

Equation 5

$W = I_1 \times A_1 + I_2 \times A_2 + I_3 \times A_3 ..... + I_n \times A_n$

where:

W = sound power in watts,
I_n = intensity for the nth surface area,
A_n = nth surface area.

Sound power can be converted to sound power level by expressing it in decibels:

Equation 6

$L_w = 10 \text{Log} [W/W_o]$

where:

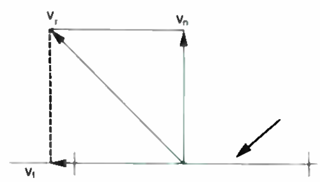


Figure 2: Intensity Components

L_w = sound power level,
W = sound power,
W_o = reference sound power, 10^{-12} watts.

When dealing with intensity and power, it is important to recognize that the power is computed based on the component of intensity which is normal (perpendicular) to the surface area. This is shown in **Figure 2**. The resultant intensity vector V_r has two components, V_n and V_t which are the normal and tangential components respectively. Power for this nth segment is computed by multiplying the magnitude of the normal component times the associated surface area, i.e. $V_n \times A$.

To develop an understanding of how these three quantities are interrelated, consider a simple noise source such as an omnidirectional loudspeaker. Equation 5 is the power response of it. Assume that the speaker has an output of 1 watt. This speaker, by definition, has a sound power level of 120 dB relative to 10^{-12} watts. If the speaker radiates sound in the form of spherical waves, then the sound pressure level varies directly with distance where the strength of the wave-front is inversely proportional to sound power level. This is referred to as a point source, a monopole source, which is expressed mathematically as:

Equation 7

$L_p = L_w - 20 \text{Log} (r) - 11,$

where:

L_p = sound pressure level,
L_w = sound power level,
r = distance in meters.

At one meter, the speaker has a sound pressure level of 109 dB. At two meters, L_p is 103 dB, a decrease of 6 dB. At four meters, the level is 97 dB, a decrease of another 6 dB. What is the trend?

As distance increases, the sound pressure level decreases at a rate of 6 dB per doubling of distance but the sound power level remains *constant*. Why? Because power is invariant with distance! Intensity decreases at a rate of 3 dB per doubling of distance. For a simple monopole source, as the area increases, the intensity will decrease at the same rate which will result in a constant sound power.

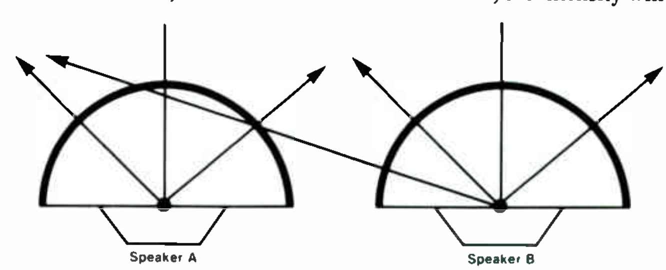


Figure 3: Contaminating Noise Source

Room Equation

Why are these important? Equation 8 shows how these concepts are used in practice.

Equation 8

$L_p = L_w + 10 \text{Log}[Q / (4 \times 3.14 \times r^2) + 4 / (S \times \bar{a})]$

where:

L_p = sound pressure level,
L_w = sound power level,
Q = directivity,
r = distance from the source, meters,
S = surface area of the room, sq. meters,
ā = average absorption of the room.

Sound pressure level is calculated as a function of the (1) sound power level of the source, (2) distance from the source, (3) direction from the source, (4) room area, and (5) average absorption of the room surface. For a specific room, direction, and distance from the source, L_p is directly proportional to L_w but generally L_p is a function of power, direction, distance, area, and absorption.

How does this relate to the loudspeaker? If the speaker has a sound power output of 1 watt, i.e. L_w of 120 dB, and the room is acoustically hard, e.g. ā = .05 and the surface area is 1,000 square meters, the sound pressure level can be computed as a function of distance. At one-quarter meter, L_p = 121 dB. At one-half meter, L_p = 116 dB. At one meter, L_p = 112 dB. At two meters, L_p = 110 dB. At 10, 20, or 40 meters, L_p = 109 dB. At small distances, e.g. one-half and one-quarter meters, the level drops at a rate of 6 dB per doubling of distance but it converges to a constant sound pressure level in the reverberant field for large distances, e.g. 10, 20, or 40 meters.

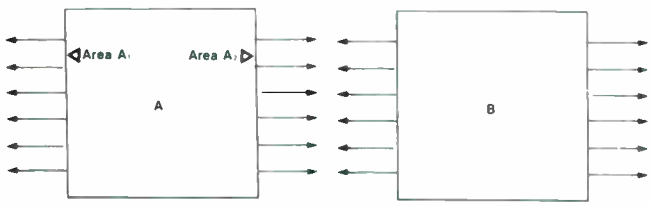


Figure 4: Plane Wave Noise Sources

Equation 8 explains why sound power is important. When power is known, sound pressure level can be computed for any spacial location in a room.

In the past, equation 8 was not used because L_w was not available for most sound sources. L_w had to be measured under highly controlled laboratory conditions which made it impractical to obtain sound power. With recent advances in technology, power can be measured in the field using the "Sound Intensity Method."

Sound Intensity Method

What is it? It is an old idea which has come of age with modern computer technology. The idea of using two microphones to measure intensity has been in existence for 40 years. Past attempts have had limited success. The crux of the method is to estimate velocity via the pressure difference between two phase matched microphones and their respective systems. Unfortunately, phase has always been a major problem in the past. With modern spectral analysis equipment (FFTs) and phase matched systems, it is possible to measure intensity with reasonable accuracy and precision.

Intensity is measured in the direction along the axis of two microphones using the "cross spectral density function." The imaginary part of this function is directly proportional to net sound intensity. In equation form, the sound intensity is:

Equation 9

$$I(f) = P_1 \times P_2 \times \sin(\text{phi}_{12}) / (4 \times \text{rho} \times dr \times \rho \times f)$$

where:

I (f) = intensity at frequency f,
P_1 = mean square pressure of microphone one,
P_2 = mean square pressure of microphone two,
dr = distance between microphones one and two,
phi_{12} = phase angle on the acoustic signal,
f = frequency.

Sound power is computed from intensity by integrating intensity times area. Subject to reasonable limitations, the power output of a source is determined with confidence

(continued on page 53)

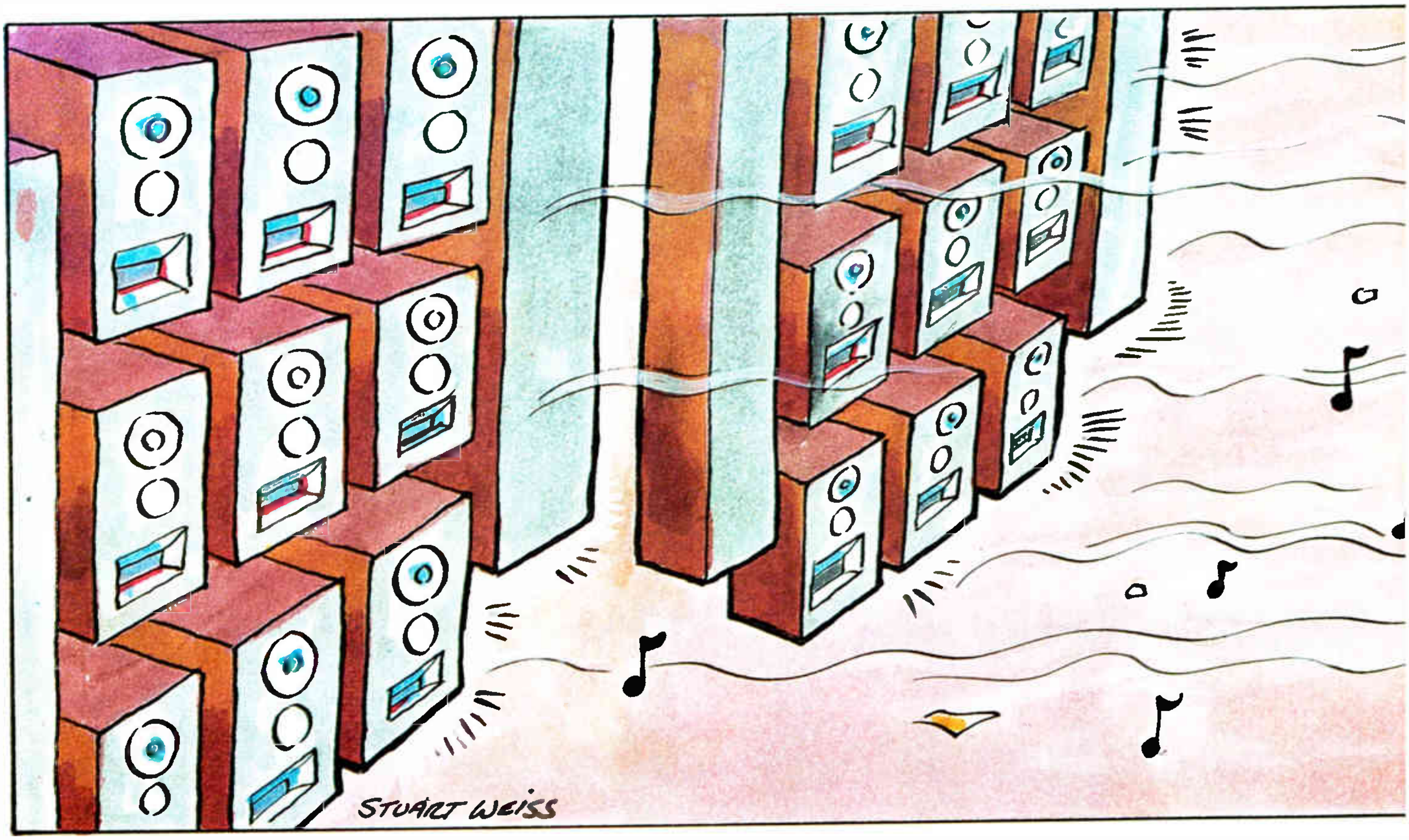
From Smoke to Chips

Audio and acoustical analyzers are familiar to the sound contracting world, as is digital-audio. However, what is new are highly sophisticated digitally-based analyzer systems for personal computers. Until a few years ago, computerized FFT analyzer systems were available only to large research organizations. Digital FFT spectrum analyzer systems are now found next to the toolboxes of acousticians, sound engineers, and sound contractors.

250 Years Of Acoustic Analysis

Scientific analysis of sound dates back to Pythagoras' investigations of the vibrations of strings. However, the science of sound as we know it really began in earnest in the 1700s. Of the many contributing researchers during the 18th and 19th centuries, the most notable were Fourier, Helmholtz, and Rayleigh. Fourier's contribution was his mathematical description of a complex waveform; often referred to as the "Fourier's Series," or FT (Fourier Transform). Helmholtz used resonators to analyze musical instruments in terms of fundamental frequencies and their harmonics. Rayleigh analytically described the vibrations found in nature. It is often amusing that many so-called modern inventions find their roots, or even descriptions, in his two-volume book, *The Theory of Sound*.

In the early years, the scientists' interests were focused on taking "pictures" of sound waves. Edison's invention of the phonograph in 1877, and the Scott-Koening phonautograph in 1859, provided photographic means of displaying sound waves. A method based on the work of Toepler, was developed and subsequently used by Mach, Wood, Foley, Souder, Sabine, and Knudsen, by which instantaneous photographs can be obtained of the



e and Sparks and Bits

by Jesse Klapholz

propagation of sound waves through scale models.

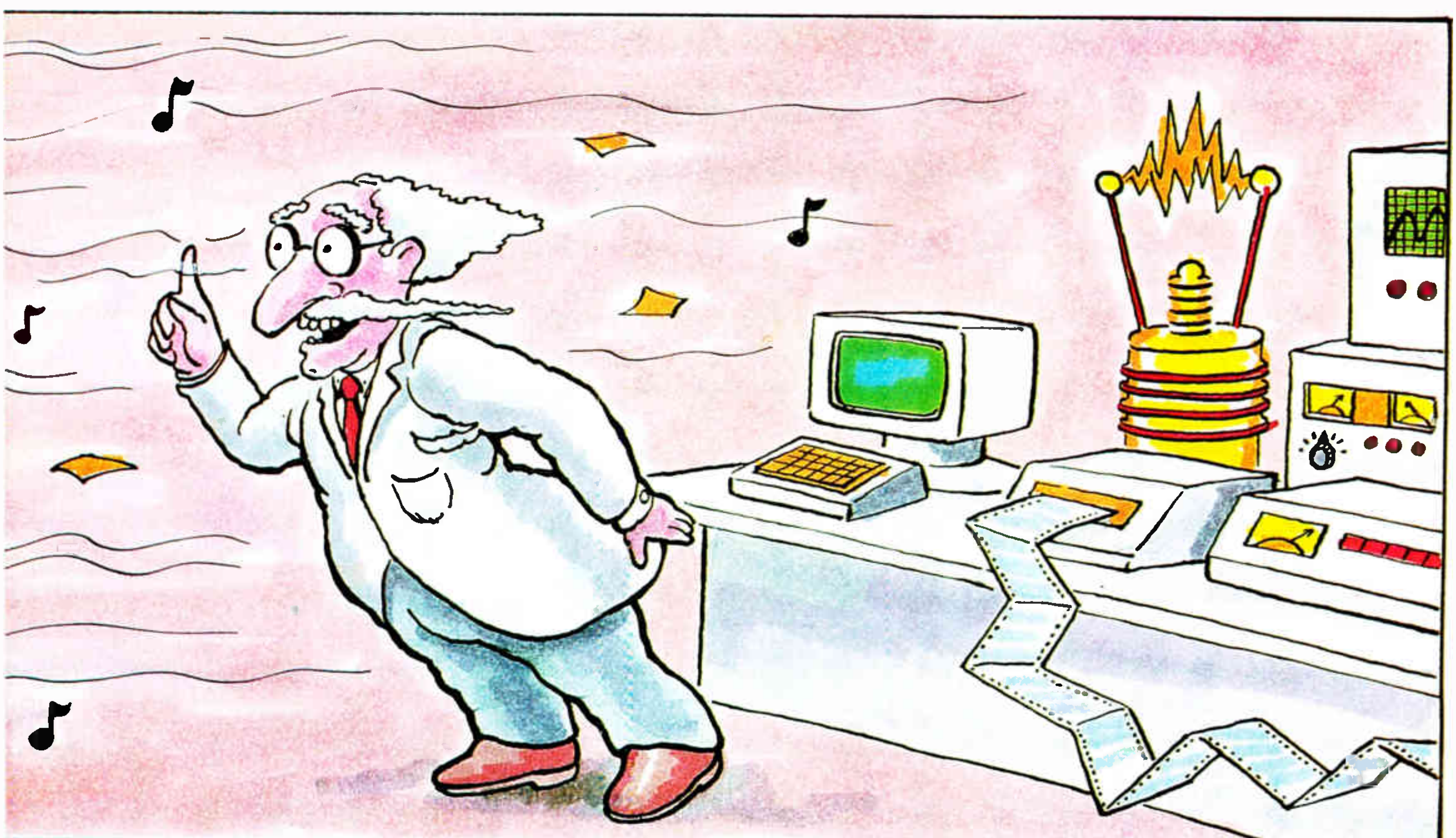
The telephone invented by Bell in 1876, and Hugh's microphone transmitter in 1878, enabled sound waves to be converted into a corresponding electrical signal. These electrical signals were first easily made available for visual inspection by the oscillograph in 1893 by Blondel and later by Dudell in 1897.

While all of these techniques (with the exception of Helmholtz's work) were concerned with taking "pictures" of sound waves, the harmonic content of the captured sound was important too. The process of analyzing a sound wave's curve consists of finding the particular numerical values of the coefficients of the Fourier equation so that it will represent the given curve. Fourier showed how this could be done by calculation (more on this later), but as it was a long and tedious process, requiring perhaps several days of work for a single curve, various mechanical devices were developed to hasten the process.

Subsequently, Henrici's Harmonic Analyzer in 1894; Miller's Amplitude and Phase Calculator in 1916; Michelson's Harmonic Analyzer and Synthesizer in 1898; and planimeter type of harmonic analyzers by Rowe in 1905, and Chubb in 1914, all reduced the calculations of the Fourier Series to a matter of minutes. It was the advent of vacuum tube technology in the early 1900s that marked the beginning of modern analysis, advancing the field from mechanical analyzers to electronic machines.

Modern Analysis

Dating back as far as the beginning of modern analysis, the problem of poor correlation between subjective quality and objective frequency



response measurements was of great concern. Shorter at the British Broadcasting Corporation, developed a technique by which decay spectra could be presented in a three-dimensional display representing amplitude, frequency, and time. During World War II, R.K. Potter at Bell Labs was working on a revolutionary new frequency analyzer that gave a continuous (real-time) analysis/display called "visible speech."

In the early 1950s, Olson, Preston, and May, working on loudspeaker development at RCA in Princeton, NJ, refined tone-burst testing of loudspeakers. Concurrently, at RCA's Camden lab, Corrington and Kidd designed a device to measure, as a function of frequency, loudspeakers excited by apodized tone-bursts.

Problems of instrumentation dominated the situation for some years. Earlier methods suffered from poor signal-to-noise ratios and were tedious. The results were also difficult to interpret and none of the techniques described came into general use. Investigations into phase distortion in loudspeakers were carried out by Wiener as early as 1940, but he was unable to eliminate the effects of the linear phase-shift induced by the transit time of the signal from the loudspeaker to the test microphone. Subsequently, Ewaskio and Mawardi measured group delay and succeeded in eliminating linear phase-shift. Later still, Stroh used a delay line for the same purpose.

Contemporary Analysis

The problems of phase-shift in loudspeaker measurement were then solved, using analog methods developed by Richard C. Heyser in 1967. Heyser's Time Delay Spectrometry technique or "TDS" (as implemented with a spectrum analyzer and oscillator) is described as: the external oscillator introduces a time offset equal to the transit time of the test signal (the time it takes the signal to travel through the air from the loudspeaker to the test microphone), and delays the tracking filter in the spectrum analyzer proportionately, thus eliminating the linear phase-shift distortion problems encountered in previous methods.

Bruel & Kjaer consequently introduced a "Time Delay Spectrometry Control Unit," which enabled those who already owned a B&K heterodyne analyzer setup to use the "TDS" process. Later, Crown International (Techron) introduced a dedicated test instrument using TDS techniques.

On the digital side, Manfred Schroeder, working with his colleagues at both: Bell Labs in Murray Hill, NJ, and at the University of Gottingham, in Germany, used a computer for acoustical analysis of enclosures. They combined signal generation, evaluation of acoustic data, and plotting of the results in one general method. The computer processing also allowed Schroeder to evaluate reverberation times based on different portions of the sound decay, energy of direct sound, early- and late-arriving reflections, and directional distribution of early sound.

Based on Schroeder's work, in 1971, impulse-response testing at KEF Electronics was being used to show steady-state and transient response and, in addition, present total system information in ways which communicated more visual information about loudspeaker behavior.

The Time and Frequency Domains

The Time and Frequency domains enable us to view physical phenomena from different perspectives. By changing perspective from the time domain to the frequency domain difficult problems can often become clear. The information content is the same in

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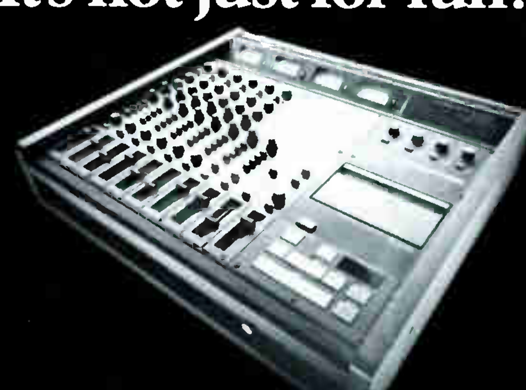

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both domains.

The traditional way of observing signals is in the time domain, which is a record of what happened to a parameter of the system versus time. Converting the parameter of interest to an electrical signal using a transducer, then showing its displacement on an oscilloscope, is an example of time-domain analysis. We say that changes in this displacement represent the variation of some parameter versus time.

Another way of representing the variation of these parameters is in the frequency domain, normally shown as a "spectrum" display. A spectrum is a relationship usually represented by a plot of the absolute or relative value of some parameter versus frequency. Every physical phenomenon, in whatever system, electromagnetic, mechanical, thermal, etc., has its own unique spectrum.

The correspondence between every time-domain function (signal) and a specific frequency-domain function (spectrum) was first established by Fourier; thus, the frequency-domain plot of a signal is called its "Fourier spectrum." Fourier's theorem states that any signal can be expressed as the sum of a component and a number (from one to infinity) of sinusoids the relative phases and frequencies of which sum to the magnitude and polarity of the original signal. The Fourier transform technique became popular in the 1960s when researchers developed a mathematical shortcut, the FFT (Fast Fourier Transform), minimizing the computation time. Subsequently, manufacturers of test equipment introduced FFT-spectrum analyzers which became an integral part of the measurement industry.

The Fourier transform technique became popular in the 1960s when researchers developed a mathematical shortcut, the FFT (Fast Fourier Transform), minimizing the computation time.

Perhaps the most straightforward approach to spectrum analysis is to present the time-domain signal to a bank of narrow-bandpass filters—each of which is tuned to a different frequency. If we then added a meter to the output of each filter, we could display the power in the portion of the spectrum passed by the filter. This can be in the embodiment of one-third-, or one-octave real-time analyzers, or FFT spectrum analyzers.

The FFT Analysis System

The solid-state and computer era provided the means to implement Fourier's and Helmholtz's theories, and brought them into our laboratories and toolkits. The technique of the FFT starts with a digitized signal waveform, which is actually a list of numbers. Using micro-computer number-crunching, we transform the time-domain list made from the original signal into a second list of numbers, one from each of the frequencies we have analyzed. The resultant list of numbers is then displayed on a CRT in the form of a graph.

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PCs for FFTs

Recently, with the influx of personal computers in the test and measurement world, an increasing number of FFT systems are being introduced that perform spectral-analysis using plug-in cards for both IBM and Apple computers. These instrumentation systems can accomplish FFT analysis of physical systems or analyze arbitrary signals for power spectrum, phase, and group delay characteristics. System facilities provide for test signal generation, data acquisition, analysis, storage, and plotting of real-time wave forms and spectra in either the time or frequency domain or three-dimensionally in both domains simultaneously.

A useful feature of microprocessed FFT spectrum analysis is that any part of the captured waveform can be analyzed, i.e. the initial attack/transient, the steady state, and the decay tail or any combination of these. Data may be presented as a fundamental frequency and its harmonics, and may also show how all of these components take place in time. We can view sampled sounds "jumping" from one domain to the other, gathering information that can be used for investigative purposes, fine tuning of instrument construction, or building up "wave shape tables" for digital speech synthesis.

System software features versatility and ease-of-use while clear presentation of data is provided by PC high-resolution graphics. These features are valuable tools for "before" and "after" pictures, various comparisons between stored-on-disc information and device(s) under test, etc. As DSP (digital signal processing) power grows and its cost decreases, we will soon see microprocessed FFT spectrum analyzer systems at the same cost/power-point as hand-held scientific calculators.

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How Impulse Testing Works

by Bill Lobb

One way of measuring the acoustical qualities of a room is by using an impulse—a narrow sliver of sound such as a click or a bang. A sound such as this is over in about a millisecond, so everything that happens beyond that millisecond is the room's response to the sound and can be observed with a microphone and display device.

How A Good Impulse Can Be Generated

Probably the simplest example is a bursting balloon. The balloon contains air under pressure, far greater than the ambient air pressure in a room. When the balloon breaks, it suddenly increases the air pressure outside the balloon. Since there is no more air coming from the balloon, the pressure falls back toward the ambient level, undershoots slightly from the momentum, and then settles back to the ambient level. **(Figure 1a)**

This change of air pressure is called an N wave because of its shape, and it is characteristic of any explosion. When viewed with a larger time scale, the N wave looks like the ideal impulse we are seeking. **(Figure 1b)**

Figure 1: Basic Impulse Waveform

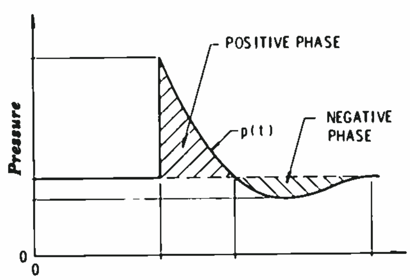

Figure 1a

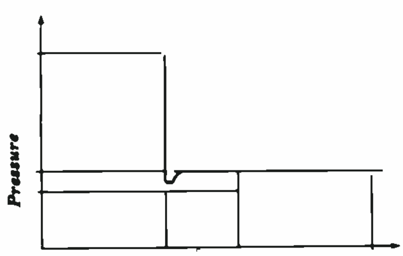

Figure 1b

The impulse in **Figure 1b** is the way a room sees it since a room has a time constant of many milliseconds, perhaps thousands of milliseconds. Fourier transformation of an impulse shows that it has a flat frequency spectrum, so we can expect that a room is being excited equally at all frequencies when subjected to an impulse.

Over the years, acousticians have improvised various devices for generating the N wave, including: large balloons, pistols, shotguns, yachting cannons, beer can launchers, spark discharges, and pulsed loudspeakers.

The requirements for a good impulse source are that it be loud, that it produce a wide frequency spectrum, that it be omnidirectional, and that it be highly repeatable in these characteristics from one shot to the next. Of the devices mentioned, only the pulsed loudspeaker is perfectly repeatable from shot to shot. And only the spark discharge and pulsed loudspeaker can be precisely triggered when it is necessary to synchronize the impulse with an oscilloscope or digital acquisition system.

It could be argued that a pulsed loudspeaker is not a truly explosive source, since the speaker itself is a mechanical system with an impulse response of its own. For small loudspeakers, however, the time constant is only a few milliseconds which is small compared to the room being measured. So, the loudspeaker appears to be a good candidate for a repeatable, easily triggered impulse source.

As for the requirement for omnidirectionality, two four-inch speakers can be configured to be ominidirectional in the following way. **(Figure 2)**

In **Figure 2a** a single four-inch speaker is shown mounted in a minimal box. Below approximately 300 Hz the box presents no obstacle to sound so the radiation is omnidirectional. Above 300 Hz the radiation is hemispherical. **Figure 2b** shows two four-inch speakers back to back in a minimal box. If the microphone were located to the right of the figure, it would hear only one speaker above 300 Hz and both speakers below 300 Hz. This is a fortunate coincidence since below 300 Hz the output of the system drops off due to the small box volume.

The result is a fairly good approximation of an omnidirectional source from 100 Hz to 4 kHz which is in the range of interest for most measurements of this type. **(Figure 3)**

One might think that two four-inch speakers could not possibly generate a big enough bang for impulse testing in large rooms. This is not so. Because of the short duty cycle requirements of impulse testing, the speakers can take an unbelievable wallop without fear of damage. Instead of using an amplifier, it is more convenient to discharge a 20 uF capacitor through the speakers with a triggered triac. The pulse width would be about equal to the time constant (20 uF times 4 ohms) or .1 millisecond. A 170-volt charge on the capacitor produces enough sound level for a large auditorium and a discharge of this magnitude doesn't seem to bother the speakers at all. The peak

For years, the loudspeaker voice paging marketplace was treated like an unsophisticated after-thought to internal communication systems. At Harris/Dracon we began changing that approach when we introduced the industry's first multi-featured, expandable, factory prewired system in 1976.

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Circle 253 on Reader Response Card

Figure 2: Approximation of an omnidirectional loudspeaker

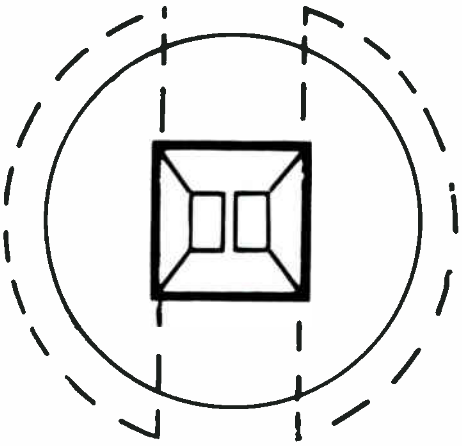

Figure 2a

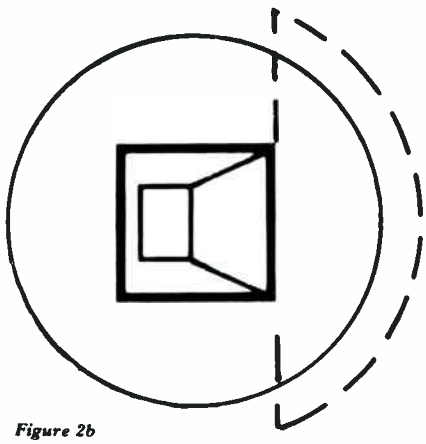

Figure 2b

Figure 3

level at one meter in front of the array has been measured at 128 dB, SPL. Interestingly, producing a pulse of this magnitude through the speakers would require the services of 3,500 watt amplifier.

Figure 4 shows the impulse response of a concert hall. The loudspeaker pulse source is on the stage and the omnidirectional microphone is located at a typical seat in the hall. The pulse source has also triggered the display, so the line of inactivity from the left of the trace to the first pulse represents the distance from the pulse source to the microphone, in this case about 35 milliseconds or 40 feet. The first spike is the direct pulse from the speaker, everything else is a reflection of this pulse coming from some surface in the hall. Clearly, there is a lot of information here, the question is what does it all mean?

Unfortunately, there has been very little work done to explain exactly what is going on in a picture like this. The particular display of **Figure 4** was digitized and stored in computer memory so the amplitude and arrival time of each spike is precisely known. Furthermore, the amplitude value of each spike can be manipulated and placed in the display in various ways. For instance, if each value were squared, we would have a picture of energy since energy is equal to amplitude squared. Squaring would cause all the negative spikes to go positive because the square of a negative number is positive. Also, the squared values could be logged to give a vertical log scale instead of the linear scale of **Figure 4**.

Further, we could make A/B comparisons by changing some acoustical condition before taking a second shot. The computer could then subtract the values of the first shot from those of the second, producing a display of the differences caused by the physical change. As a matter of fact, A/B comparisons have proven to be one of the most valuable aspects of impulse response testing, especially for those interested in modifying the acoustics of rooms, either electronically or physically.

Other possibilities of digital processing are: integration, averaging and smoothing, and calculation and display of the reverberation decay slope.

If the original impulse was positive, why do there appear to be just as many negative going spikes in the picture. Well, each time a sound is reflected from a surface its phase is changed by 180 degrees. A positive pulse that has undergone one reflection is negative, on the second bounce it is positive again, and so on.

The number of bounces is characterized by the "order" of the reflection: one bounce, first order; two bounces, second order, etc. Obviously a 26th order reflection will arrive at a much later time than a fourth order one. So, all the positive going spikes are even order reflections and all the negative going spikes are odd order reflections. Armed with this information and some scale drawings of the room, it becomes an easier task to find the source of a reflection of interest.

Since the amplitude of reflections must diminish as time goes on, how is it that reflections are often seen that equal or occasionally exceed the amplitude of the direct pulse? The only answer I can think of is that these later large reflections are not one, but a coincidence of reflections arriving at the microphone in phase. Although it is possible that a number of, say 10th order reflections, could arrive at the microphone from different parts of the room at the same time, it is statistically unlikely. It is more likely that the large reflection came from a 90 degree corner or a curved concave surface where a considerable amount of energy can be gathered and shot back at the microphone. Large lonely spikes in a field of decaying randomness are the stuff of which echoes and bad sound is made.

Digital Processing

Because of the mind boggling complexity of a room's impulse response, most researchers content themselves with evaluating only the gross characteristics of the display. However, digital processing is now making it possible to concentrate on a particular reflection and to find out where it came from and what part it plays in the perceived sound of the room. This is the goal of impulse testing.

When evaluating an impulsed room's signature, certain general characteristics should be looked for. Generally a line can be drawn through the display at about 200 milliseconds after the direct sound pulse. Everything to the right of that line is reverberation and is heard as a "tail" on any sound produced in the room. If there are any tall spikes in this region, they are echoes with possible serious consequences.

Everything to the left of the line, up to the direct pulse, represents "early"

Isolating Sound for Measurement

by **Bruce Bartlett**
Crown International

It's a new age for the sound contractor. Acoustic measurements that were never before possible can be performed by computer-age test equipment such as the Techron TEF System 12, a portable computer designed to make quick, accurate measurements of room acoustics and sound systems.

We're used to seeing test gear with an array of knobs and switches. The TEF System is different; there are no knobs to set. Instead, all control settings are done through the keyboard (aided by prompts from the built-in monitor screen). The advantage is that settings can be recalled and exactly duplicated whenever needed. In addition, the computer becomes whatever piece of test equipment the software tells it to be—increasing its flexibility.

How It Works

This sophisticated instrument is the hardware embodiment of Time Delay Spectrometry developed by Dr. Richard C. Heyser. The TEF machine generates a frequency sweep into a sound system, then picks up the sound of the sweep through a microphone. The microphone signal is fed through a filter that tracks the sweep. This tracking filter can be time-offset to compensate for the travel time of sound from speaker to microphone. By varying the bandwidth and time-offset of the tracking filter, you can study the spectrum of the direct sound by itself, certain sound reflections, or both.

The tracking filter also increases the signal-to-noise ratio of the measurement by filtering out frequencies other than the one being measured. Consequently, accurate tests can be run even in noisy environments, with conversation going on in the background.

Some other time-selective measurement methods that eliminate the need for anechoic conditions are the gated FFT, tone-burst, impulse, and cross-spectrum methods. The TDS technique offers superior signal-to-noise ratio because it uses a higher-energy test signal and filters out background noise.

The TEF System 12 can measure energy vs. frequency (frequency response), energy vs. time (energy level of sound reflections vs. time), and frequency response vs. time (3-D display). It also will do phase measurements and Nyquist plots. Measurements made at different times or places can be compared and differenced.

Typical Application

With the TEF System 12, you can actually measure the anechoic frequency response of a speaker cluster after installation. The analyzer can remove all the room reflections from the measurement, leaving only the direct sound. Or it can show the effect of early sound reflections on the speaker-system response, excluding the room reverberation. This information is not available on an RTA display.

The ability to separate direct sound from the total sound field is essential, for it is the spectrum of the direct sound (and early reflections) that determines the perceived tonal balance of a speaker system. In other words, only the direct sound and early reflections contribute to the perceived tonal balance. The long-term reverberant spectrum as shown on an RTA display does not correlate well with listening tests.

Acoustical consultants use the TEF System 12 to focus on acoustic trouble spots such as echoes, early reflections that cause comb filtering, and so on. They can make in-situ measurements of the absorption vs. frequency of acoustic treatments.

The TEF System 12 makes audible phenomena become visible on the screen. For example, the TEF analyzer can show the pattern-in-time of sound reflections in a room. **Figure 1** shows a typical display of the energy level of sound reflections vs. time. The tallest line to the left is the direct sound, followed by discrete early reflections, followed by closely spaced random reflections or reverberation. When you place a cursor on a spike indicating a reflection, the screen displays the arrival time of that reflection.

If a strong cluster of reflections oc-

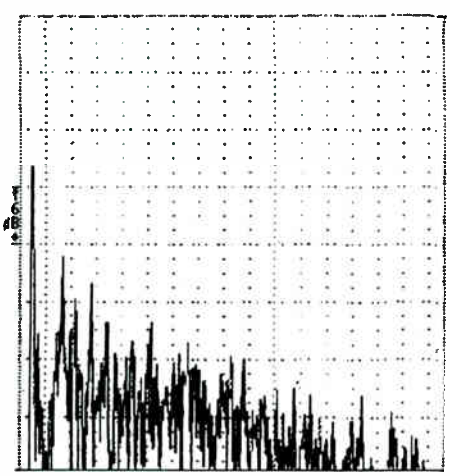

Figure 1: A typical display of the energy level of sound reflection vs. time.

curs more than 50 milliseconds after the direct sound, this creates an echo which can impair intelligibility. The TEF System 12 lets you determine the arrival time and, hence, the source of these reflections. Once the problem reflections are identified, the offending surface can be modified to diffuse or absorb the incident sound. You don't need to acoustically treat the entire room because only those surfaces causing the problem need be treated. This can save the expense of unnecessary modifications.

The TEF System 12 can be used as a regular computer as well. It includes three Z-80 microprocessors which let you run CP/M or BASIC programs such as word processing, circuit analysis, or sound-system design.

TDS Theory

How does Time Delay Spectrometry work in a speaker-measurement arrangement? The TEF analyzer generates a sine-wave frequency sweep which is played through a loudspeaker. The change in frequency is linear with time. The microphone picks up the sound generated by the loudspeaker, and the microphone signal is filtered by a bandpass filter that tracks the generated sweep.

The tracking filter is not in sync with the generated frequency sweep. It is time-offset to compensate for the propagation delay of sound traveling from speaker to microphone. For example, at the instant a 1,000-Hz tone reaches the microphone through the air, the tracking- filter center frequency is set to 1,000 Hz.

Now suppose that the loudspeaker's sound reflects off a wall and enters the microphone after a certain delay. By the time the reflection enters the microphone, the tracking filter will have swept to a higher frequency than the reflection (**Figure 2**). If the filter bandwidth is sufficiently narrow, the reflection is rejected or filtered out. No reflection signals are received by the TDS analyzer. In other words, an anechoic measurement has been made in an ordinary room.

The bandwidth of the tracking filter can be preset. The wider the bandwidth, the greater the "time window." This is a range of time over which signals are accepted by the analyzer. The relation between time window, bandwidth, and sweep rate is T = B/S, where T = width of time window in seconds; B = bandwidth in Hz, and S = sweep rate in Hz/sec.

Since sound travels a certain distance within a time interval, the time window corresponds to a "space window." The space window is an ellipsoid space around the speaker and microphone, inside of which sound reflections are included in the measurement. The speaker and microphone are at the focii of the ellipsoid (**Figure 3**). Sound reflections originating outside the space window are excluded from the measurement. Actually, they are attenuated 3 dB at the edge of the space window ellipsoid, and by greater amounts outside that.

On the TDS analyzer, the space window is determined by setting the bandwidth and sweep rate. A 10-foot space window coresponds to a bandwidth setting of 88.5 Hz at a sweep rate of 10,000 Hz/sec. Here is the appropriate formula: B = SD/C, where:

B = bandwidth setting of tracking filter in Hz
S = sweep rate in Hz/sec
D = space window in feet
C = speed of sound, 1,130 feet/sec

The larger the space window, the lower the frequency that can be measured accurately. That is, the lowest frequency of resolution decreases as the space window increases. Therefore a relatively large, empty room is needed for low-frequency measurements.

The relation between resolution frequency and space window is F = C/D, where:

F = resolution frequency in Hz
C = speed of sound in feet-per-second
D = space window in feet.

If we want to measure down to 100 Hz, we need a space window of roughly 10 feet, or a clear space five feet around the microphone and loudspeaker (from the formula D = C/F). Using graph paper, check that the path length of each room reflection exceeds the direct-sound path by more

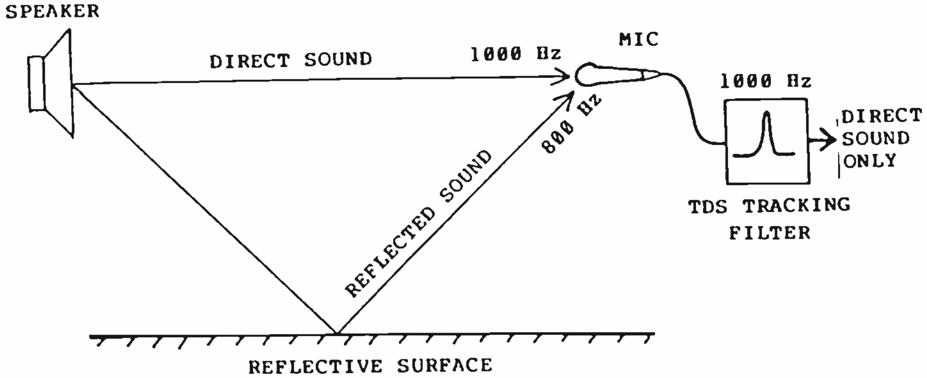

Figure 2: By the time the reflection enters the microphone, the tracking filter will have swept to a higher frequency than the reflection.

put windows of various types.

The TEF.RCH disk (TEF applications software as modified by Richard Heyser) is the same as the TEF 1.0F applications software, with these additional features:

*Harmonic distortion measurements (phase and amplitude), fundamental through ninth harmonic.

*An expanded ETC mode that lets you expand the time axis from one to 32 times. The expansion gives the ETC display a resolution corresponding to that of a 12,800 LINE FFT processor.

*FFT measurement capability from 0-25 kHz with internal anti-aliasing filter or 0-1 MHz with selectable bandwidths from 10 Hz-25 kHz.

*Oscilloscope-style display of FFT-analyzed signals.

*A special calibrated grid overlay in linear frequency, log frequency, or ISO patterns.

*Impulse response and doublet portions of ETC can be displayed independently. (Doublet is the Hilbert transform of the impulse response.)

*Group delay measurement.

*External trigger of the TDS, ETC, or FFT portions of the program. This external triggering can be used to perform a sweep on a non-synchronous incoming signal, such as a previously recorded TDS calibration sweep on a magnetic recorder.

*In extremely high-noise environments (for example, live performance), even a TEF sweep may not provide sufficient signal-to-noise ratio. The RCH disk can average multiple sweeps into a single measurement through a scalar or vector averaging method.

The Workbench Instrument Software Version 1.0 provides four instruments for bench and field testing of sound systems, electronic equipment, and other electro-acoustic systems. These instruments include a sound level meter, digital volt meter, oscilloscope, and function generator. Pull-down menus allow instant access to any of the system's parameters and setups, with full help facilities available. Functions are provided to allow storage of instrument settings and defaults, multiple printing options, and selection of input options.

Other software disks include:

V.Box.Res, a tool to aid in speaker-cabinet design, written by Don Keele.

FTC (Frequency Time Curve), a top view of the 3-D display, written by Keith Jebelian.

LINCAP, a Linear Circuit Analysis Program written by Gerald Stanley.

TEF-PEUTZ, a collection of V.M.A. Peutz software modifications including RT60 and ALCON calculations.

Various CP/M programs for word processing, spread sheet applications, etc.

Conclusion

As is true with any piece of high-tech test equipment, clients watching the Techron TEF System 12 in action are fascinated and impressed. But more importantly, they are provided an accurate analysis of their acoustic problems, which permits scientific solutions. The TEF System 12 offers surprising insights into the phenomena of room acoustics and the performance of sound systems.

Editor's Note: At the NAB and NSCA shows this spring Techron introduced its newest in TEF analyzers —the Techron TEF 12. The Techron TEF 12 replaces the Techron TEF 10.

Bruce Bartlett is a microphone development engineer at Crown International. His work at Crown has centered around the cardioid boundary microphone. Bartlett holds a B.S. in Physics from the College of Worcester, Worcester, OH, and has studied electrical engineering at Akron University. Bartlett is a member of the Audio Engineering Society.

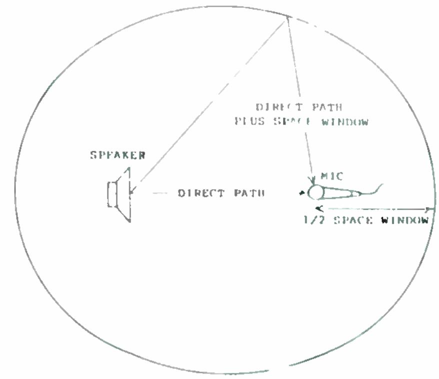

Figure 3: The speaker and mic are at the focii of the ellipsoid.

Objective Measurement of Speech Intelligibility

by E. Curtis Eichelberger
Bruel & Kjaer

A new method is now available, called RASTI (Rapid Speech Transmission Index) that allows us to rapidly and objectively measure the quality of verbal communication in auditoria, theaters, conference rooms, etc. This measure has a direct relationship to speech intelligibility. The method has recently been standardized by the IEC (IEC Publication 268, Part 16). A fully portable and self-contained system, it can carry out this measurement in as little as eight seconds and can also provide diagnostic information as to how the speech intelligibility could be improved.—ECE

Introduction

The first step in solving a problem is to quantify it. This can be quite a challenge if the problem is speech intelligibility. Many different methods have been devised in the past to measure speech intelligibility. Subjective methods have been based on "word scores" using trained speakers and a "jury" of listeners. Such methods are time consuming and are affected by human factors.

Objective methods involve the measurement of several physical parameters such as signal level, background noise and reverberation time; and these measures are then used to compute a measure of speech intelligibility, called Articulation Index (1). The major shortcoming of this method is that the desired signal and undesired noise (background and reverberation) must be measured separately. This is usually difficult, or in some situations impossible, to do. In these situations, one or more of the physical parameters required in the calculation are estimated.

In short, none of these speech intelligibility measurement methods has gained widespread use because of their complexity and subjective nature. A method called RASTI (Rapid Speech Transmission Index) (2), which is an objective measure of speech transmission, is a condensed version of the measurement method of Speech Transmission Index (3).

The RASTI method gives a single number measure of speech intelligibility ranging from 0 (Bad) to 1 (Excellent) as shown in **Figure 1.** The method automatically accounts for background noise and reverberation, and no adjustments need be applied as is the case with the Articulation Index method. Measurements are performed with both the signal and background noise present. More importantly, a measurement can be performed by one person in less than 10 seconds.

The RASTI Method

The RASTI method measures the reduction in modulation of a transmitted test signal. The test signal simulates the characteristics of the human voice that are important to speech communication: the carrier signal and the low frequency intensity modulations. The carrier signal consists of two octaves of band limited noise centered at 500 Hz and 2 kHz. The levels of these octave bands are set to the average levels found in normal speech (**Figure 2**). Although the human voice spans a frequency range of from 100 Hz to 8 kHz, 98 percent of the information in speech is in the range of 500 to 2 kHz. The low frequency modulations present in human speech are simulated in the RASTI test signal by nine discrete Modulation Frequencies between 1 Hz and 11.2 Hz. These frequencies span the range found in human speech (**Figure 3**). The modulation process creates a time varying speech envelope as illustrated in **Figure 4** for the 2 kHz octave band. An example of a human voice signal is shown in **Figure 5.**

The measurement is performed by transmitting the test signal at the speaker's location and analyzing the

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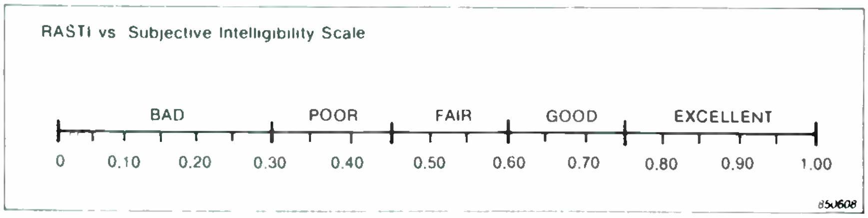


Figure 1
Qualitative interpretation of RASTI

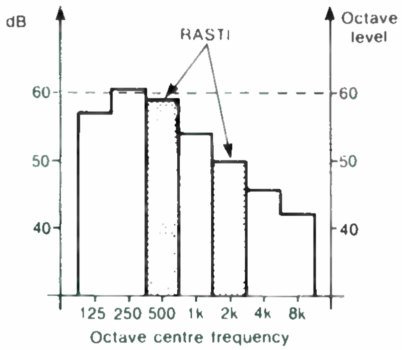


Figure 2
Average octave-band spectrum of normal speech at one meter distance. The shaded portions indicate the carrier signal used in the RASTI method.

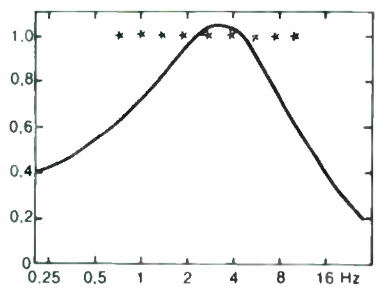


Figure 3
Curve showing the modulation spectrum of human speech. The discrete Modulation Frequencies used in the RASTI method are marked with a ★.

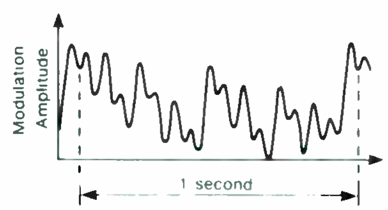


Figure 4
RASTI signal modulation (for 2 kHz octave-band).

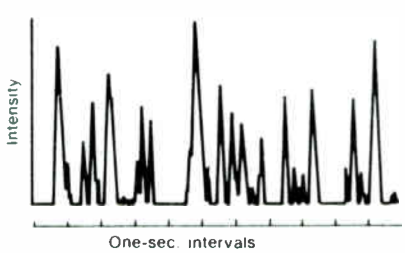


Figure 5
An example of the intensity envelope of a segment of human speech.

resulting sound at the listener's location. The analysis consists of octave band filtering and demodulation of the received signal to obtain the Modulation Index for each Modulation Frequency. Speech intelligibility is related to the reduction in modulation, or Modulation Index. **Figure 6** illustrates the reduction in Modulation Index of a simplified speech signal caused by background noise and reverberation. The modulation reduction is interpreted as though it is caused by background noise alone. The signal to noise ratios which alone would have resulted in the measured reduction in Modulation Index are calculated. The RASTI value is the arithmetic average of these "apparent" signal to noise ratios, normalized so that the index lies between 0 and 1.

Typical RASTI Applications

Projects where RASTI can be used include all locations where the intelligibility of speech is of interest. Some general applications are discussed below.

In large churches, theaters, auditoria, etc., the acoustics of the room have a large influence on speech intelligibility. The speech intelligibility can vary considerably at various locations throughout the audience. This is true regardless of whether sound reinforcement is used or not. Traditional methods of evaluating speech intelligibility, using a "jury" approach, would be impractical because of the large number of locations where measurements would be required and the time required to complete the measurements. Because of these limitations the designer of the hall usually resorts to indirect measures of performance such as reverberation time, background noise level and signal sound pressure level. These measures all interact to effect the speech intelligibility; but either one alone does not guarantee a satisfactory result.

With RASTI, the speech intelligibility could be measured by one person in a very short period of time. By means of the RASTI method, the speech intelligibility was evaluated in St. Paul's Cathedral in London (one of the world's largest churches). Measurements were performed at over 230 locations, with and without the reinforcement system, in less than four hours (4).

The rapid measurement capability of the RASTI method makes it feasible to make a complete set of measurements with and without the sound reinforcement system on; thus providing a direct measurement of the sound system performance.

In smaller rooms such as conference rooms or lecture halls, good listening conditions are of great importance. In these situations there can be several talker and listener positions. The RASTI method makes it feasible to perform a thorough evaluation of multiple talker and listener positions.

In industrial facilities that are very noisy, or in transportation vehicles such as airplanes, transit cars or busses, RASTI is an ideal method for measuring the performance of P.A. systems. P.A. systems have been evaluated successfully in power plants with background noise levels of 90 to 100 dBa (5).

One feature to look for in a RASTI measurement system is the ability to manually enter octave band levels of background noise. In this way, measurements can be made to incorporate a higher background noise than actually exists in the room under test. Using this method a P.A. system could be acceptance tested before an industrial plant is put into full operation. Another example of how this feature could be used is if the background noise is dominated by traffic noise. Listening conditions could be measured at one traffic condition and then accurately projected for other traffic conditions.

In railway stations, airports, and other buildings where reverberation is a problem, the RASTI method is an ideal tool for evaluating the intelligibility of P.A. systems.

Sound masking systems are often used in open office plans to reduce the speech intelligibility, or in other words, to increase the acoustic privacy, between work stations. The RASTI method is well suited for evaluating the effectiveness of masking (5).

The RASTI method can also be used as a preliminary diagnostic tool for intelligibility problems. If certain information is available with the measurement system, this information can then be used to decide what changes would be required to improve listening conditions. For example, if the Modulation Index (for a given octave band) decreases with increasing Modulation Frequency then we can conclude that the RASTI value is being limited by the room's reverberation (6). If the Modulation Index is low for all Modulation Frequencies then it is likely that the RASTI value is most strongly influenced by background noise (6). Differences in the effect of background noise and room reverberation to the RASTI value should also be available as a function of either the 500 Hz or 2 kHz octave bands. This information will suggest the shape of the background noise spectra or the room reverberation curve.

Summary

With the introduction of the RASTI method, those who work on the design of auditoria and sound reinforcement systems have a tool to objectively evaluate the speech intelligibility of their projects. If problems exist RASTI can also provide a preliminary diagnosis of the problem.

We expect to see widespread use of the RASTI measure in project specifications where speech intelligibility is the major concern. RASTI is suited for the performance specification. The specification of RASTI values for various locations throughout the listener seating area is sufficient to quantify the performance of a sound system.

In summary, arguments and differences of opinion abound when it comes to auditoria acoustics and the performance of sound systems. These

(continued on page 52)

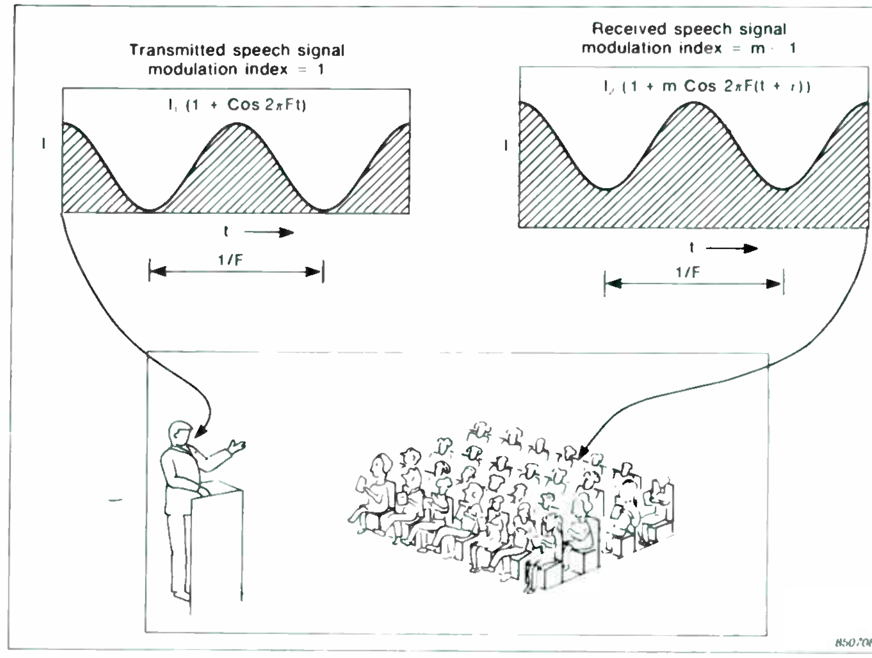


Figure 6
Illustration of the reduction in modulation of a speech signal caused by background noise and reverberation.

Live Equalizing for Performance

by Chris Michie

Figure 1

Figure 1a

Amplitude

Unequalized Loudspeaker response

Phase

Figure 1b

Amplitude

Complementary Equalizer response

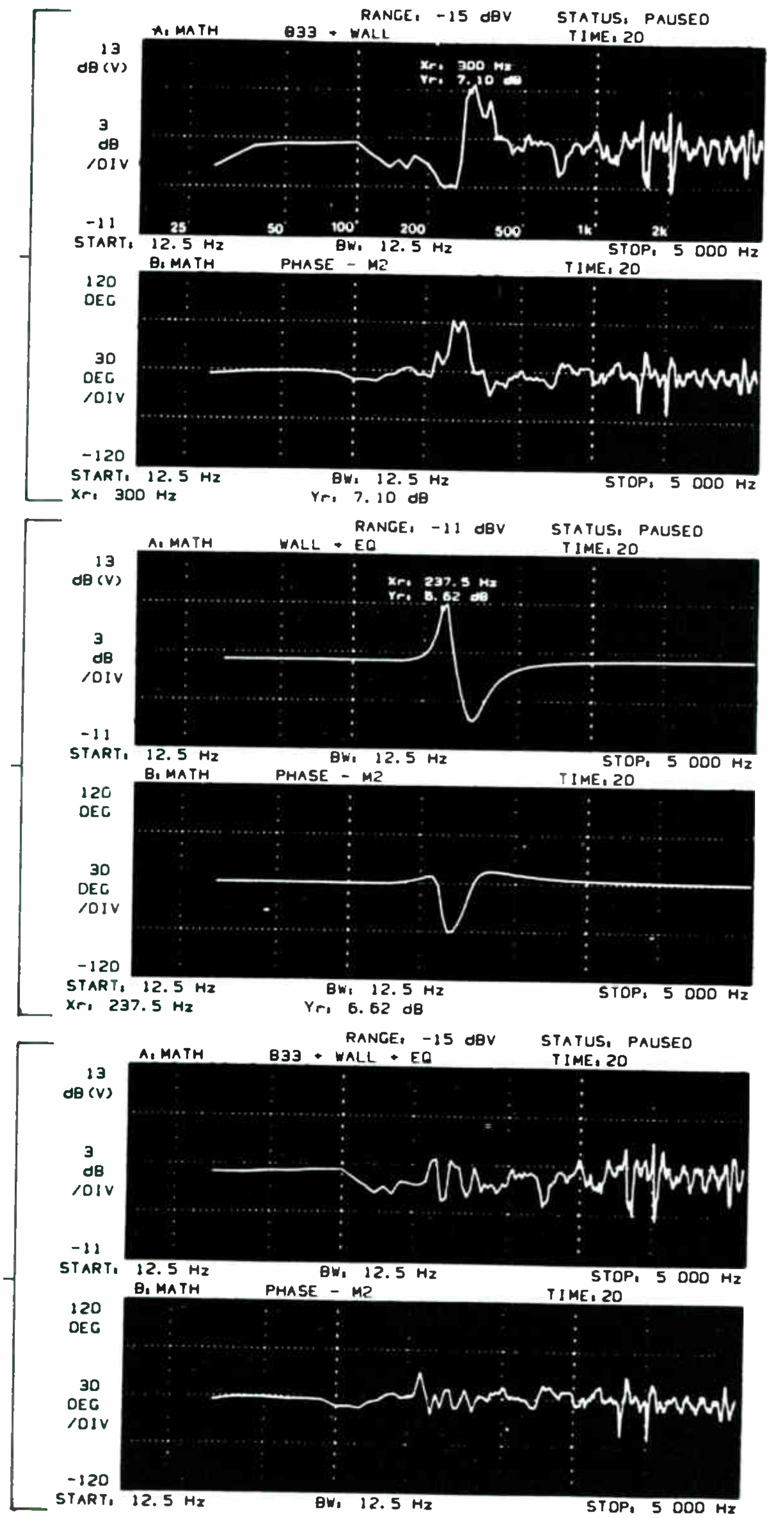
Phase

Figure 1c

Amplitude

Equalized Loudspeaker response

Phase



Examples of Complementary Phase Equalization... (from CP-10 Operating Instructions)

The Source Independent Measurement[TM] (SIM) technique is a fast and accurate method of analyzing and equalizing sound systems and how they interact with a room's acoustics. SIM allows the operator to quantify the frequency and phase response aberrations caused by the listening environment and permits the use of any test signal for the analysis, including the musical or vocal portion of a live performance. This means that accurate analysis and correction can be continued throughout a performance, with the audience present.

Using a complementary phase parametric equalizer in conjunction with a low distortion, phase-coherent loudspeaker system, the operator can effectively minimize acoustic resonances and can compensate for frequency response modifications caused by delayed reflections of up to 40 msec. The sonic improvements are dramatic, verifiable using other test methods, and repeatable in any acoustic environment.

Loudspeakers vs. Acoustics

As we are all aware, the most profound influence on the sound quality of a sound system is its environment. A system that has been measured in a free field may (and usually does) exhibit totally different frequency response and coverage characteristics when installed in a confined space. Sound system designers have learned to deal with this problem by first using real-time analysis to determine the installed system's response and then equalizing to compensate for changes caused by the environment. However, despite the secure scientific foundations of the method, the results have been less than uniformly satisfactory. Many installers have found that their system equalizers have been "retuned" by an operator to a more aesthetically satisfying setting, or disconnected altogether.

There is obviously something lacking in "traditional" methods of system equalization. Large and expensive installations that meet specifications when installed have nevertheless been modified, re-aimed, abandoned, or replaced by the new owners. The "new, improved" solution is often unsatisfactory, and the cycle continues, accompanied by much heated discussion and occasional litigation. This scenario is the result of a misunderstanding of the complexities of the problem and a reliance on inappropriate techniques.

Anti-Resonant Equalization

In order to minimize the effect of room resonances and reverberation, the sound engineer needs a device that will repair the damage caused by the accumulated delays. The sum of these delays results in measurable distortion of the frequency response of the sound system, so by inserting an equalizer in the signal chain it is possible to compensate for these distortions and regain flat response, as can be seen in **Figure 1C.** This corrective equalization may be described as "anti-resonant." However, unless the accompanying phase shift produced by the equalizer is the inverse of (or "complementary" to) the phase shift produced by the "room delay," there will be little subjective improvement in sound quality. In **Figure 1B,** this condition is satisfied.

For successful correction, it is absolutely critical to match the unwanted room resonances with anti-resonant equalization that is exactly the inverse in terms of amplitude and phase response, and identical in center frequency and bandwidth. To do this it is necessary to measure the phase and amplitude response of both the room resonances and the corrective anti-resonant equalization.

It follows that one should not apply anti-resonant equalization to a sound system without being able to monitor a corresponding improvement in the phase response. Response modification caused by delays of over 40 msec will not respond to equalization using current technology and such phase conditions must be recognizable to the operator. Unfortunately, few audio analyzers measure phase.

Obstacles to Meaningful Measurement

Compensatory equalization of sound systems in rooms, as described above, is often referred to as "room-tuning" (though de-convolution is the correct theoretical term) and there are several common non-SIM techniques. These techniques are useful in certain situations, but since all non-SIM techniques are source-dependent (usually

Figure 2

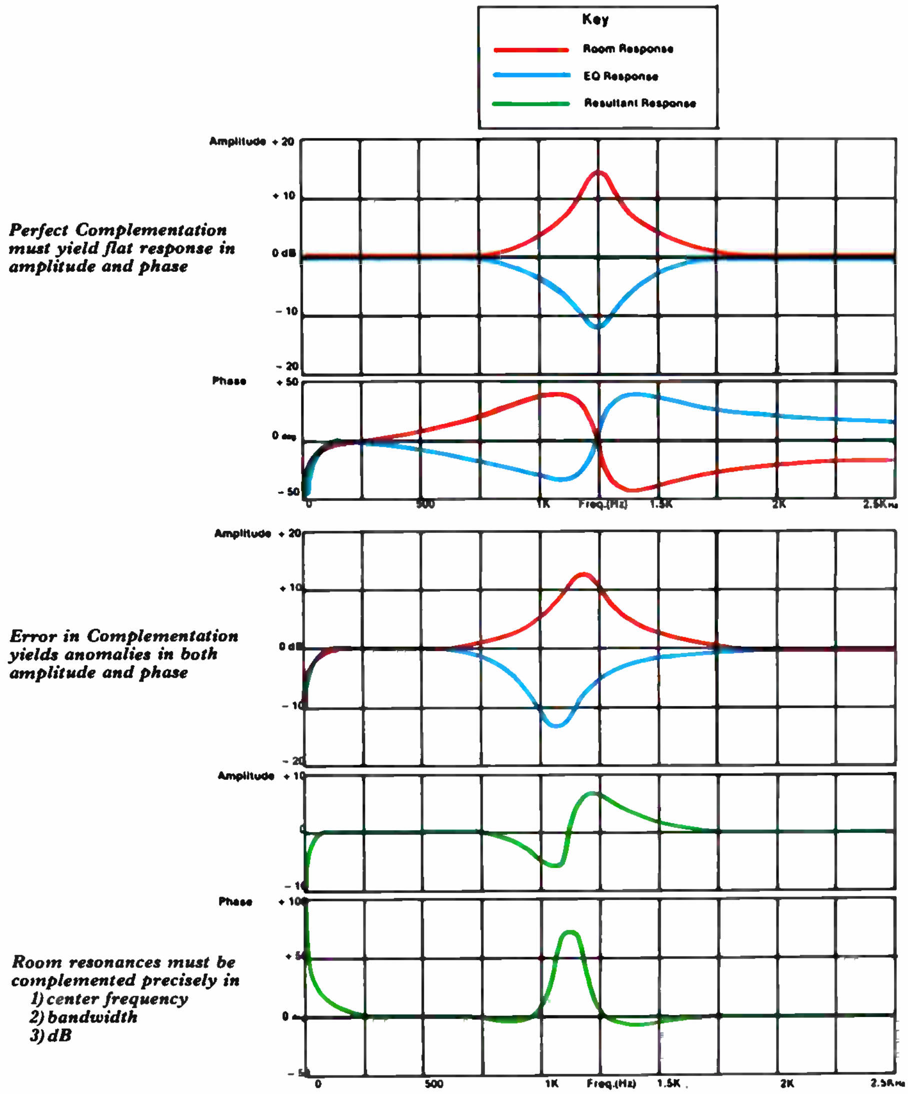


Figure 2: Complementary Equalization

Figure 3

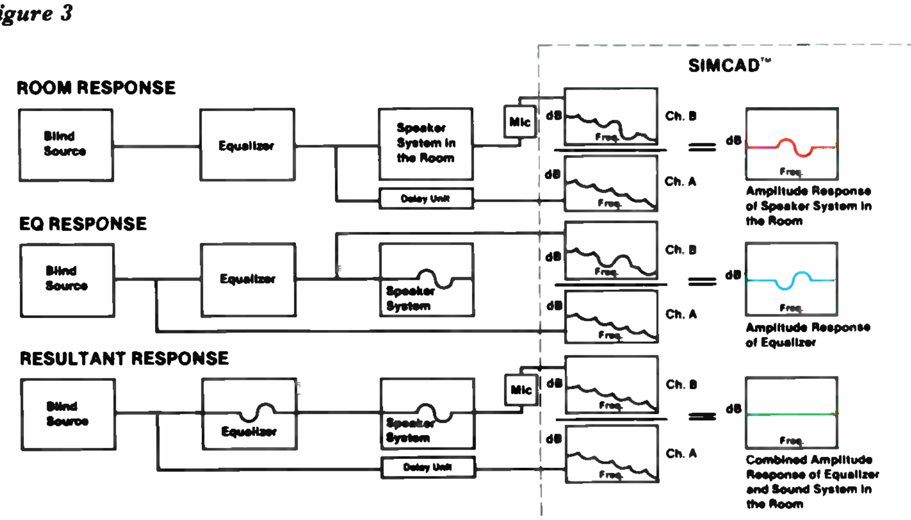
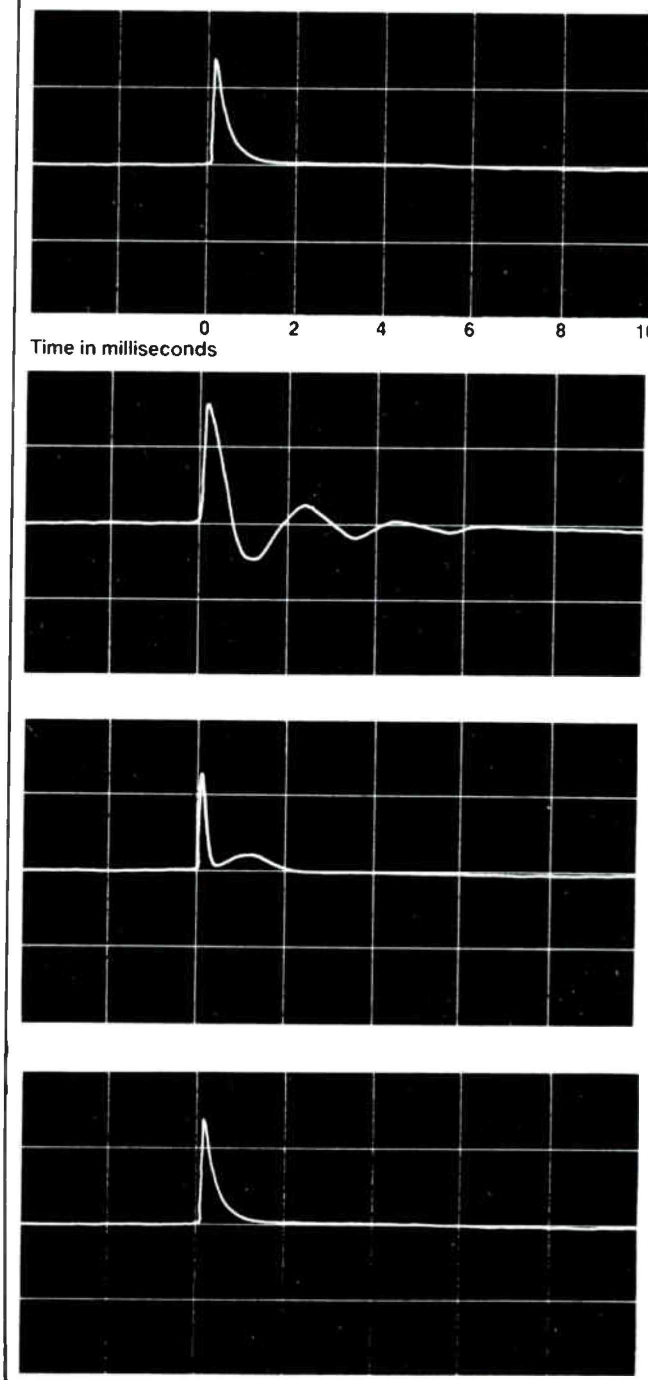


Figure 3: Source Independent Measurement of a Sound System

pink noise or sine-wave-sweep test signals) they cannot be used in most live performance situations. Attempts have been made to introduce reference signals into musical performances, but the necessary clicks and pops are clearly audible, even at levels 40 dB below program peaks, and audiences have objected. This makes it almost impossible to conduct "room-tuning" while an audience is present, and testing is therefore done prior to the performance. Unfortunately, since clothed humans represent significant amounts of highly absorbent material, results obtained in an empty hall are usually invalidated by the arrival of the audience.

Through a mathematical process called the Fourier Transform it is possible to examine level versus frequency and phase response. By using one channel of a dual-channel FFT analyzer as a reference and the other to measure the system under test, it is possible to derive the "transfer function" of the unknown system using a non-standard test signal. In other words, the analyzer can compare the reference signal with the same signal



Oscilloscope display showing four traces depicting:

Pulse (ideal impulse, pink filtered)

Pulse showing resonance due to equalization introduced in Channel One of the CP-10 (+ 10 dB at 500 Hz .3 octave bandwidth)

Pulse affected by equal and opposite anti-resonance equalization in Channel Two of the CP-10 (– 10 dB at 500 Hz .3 octave bandwidth)

Same pulse passing through both channel of the CP-10 in series, showing total correction of resonance using anti-resonant equalization. Note that there is no measurable time delay.

Figure 5: Effective equalization... (from CP-10 Operating Instruction)

after it has passed through the system under test, and display the differences in both amplitude and phase response. Any audio system or component may thus be examined while passing signal under real conditions **(Figure 3)**.

According to John Meyer, who developed the SIM technique, SIM is *not* just another implementation of an auto- or cross-correlation technique and any reference to the word "correlation" or "correlate," etc. with regard to SIM is strictly in the dictionary meaning of the word.

In live concert situations, any measurements made are subject to contamination by audience noise and other ambient interference. Noisy data segments are detected by a lack of "coherence" between excitation (the reference channel) and response (the measurement channel) and must be rejected whenever possible. Accordingly, amplitude or phase data associated with poor coherence, for example, less than .2 (where coherence is defined as being between zero and one at each frequency) is ignored when determining the necessary equalization.

This technique allows the operator to identify the non-linearities in the room/sound system output and the analyzer's digital data capture and storage technology allows for the necessary high resolution. Using an overlay function, it is possible to rehearse the appropriate equalization before insertion by exactly matching the inverse of the equalizer output to the room curve, with both curves displayed simultaneously. When satisfied as to best "fit" the operator inserts the anti-resonant equalization and verifies it effectiveness on a continuing basis throughout the performance **(Figure 4)**.

The only tool that can construct the complicated equalization curves which exactly complement the "bump-structure" in the room is a parametric equalizer, so named because it offers control of all equalization parameters (excepting phase). Since anti-resonant equalization will only be effective if accompanied by a phase response improvement, it is fortunate that the phase shifts due to resonances that occur in acoustic spaces can be duplicated electronically in a complementary phase equalizer. Unfortunately, many available parametric equalizers are essentially notch filters in the cut position, few exhibit complementary phase characteristics. By their very nature, notch filters are asymetrical or non-complementary to resonances commonly found in room/speaker environments, and their use will actually introduce a frequency-selective delay into the system.

To illustrate the practical effect of anti-resonant equalization, **Figure 5** reproduces four traces from an oscilloscope display. The first represents an ideal pulse passing through a linear system. The second shows resonance, caused in this case by equalization in one channel of the CP-10 parametric equalizer. The third trace shows equal and opposite anti-resonant equalization applied to the original pulse. The fourth trace shows, first, that the sum of the resonance and anti-resonance is completely complementary and self-cancelling and, second, that there is no net delay, despite the signal having passed through two filter circuits.

Given, then, a phase-coherent sound system, a dual-channel FFT analyzer and a complementary-phase parametric equalizer such as the Meyer CP-10, the SIM practitioner can examine and correct for many acoustic conditions using the performance material as a test signal and with the audience present. This technique not only fulfills the promises made by the original proponents of "room-tuning" as a solution to acoustic evils, but also introduces a comprehensive analysis tool for comparing any two audio signals. The applications for the system designer and installer are obvious; for the first time they can specify high-quality sound system installations that meet objective and subjective specifications with a built-in capability for modification to suit any resonable performance requirement.

Proof of the technique's validity can be found in the growing list of performing artists who have used SIM with measureable improvements to their concert sound, even in the notorious acoustic conditions of multi-purpose sports arenas. Luciano Pavarotti, The Grateful Dead, and The Thompson Twins have all used the technique to bring high-quality audio to their audiences.

As SIM is implemented to its full potential, we are seeing these corresponding results: first, discerning artists, confident that the technology exists to reliably and consistently overcome acoustic shortcomings, are consenting to perform in otherwise unuseable reverberant spaces. Second, consultants are able to recommend equipment purchases with the knowledge that the designs will meet specification and satisfy the client. Third, and perhaps most important, large paying audiences are enjoying audio quality rivaling that of the best home entertainment centers.

Editor's Note: The preceding article is an introduction to Source Independent Measurement or SIM which was developed by John Meyer of Meyer Sound in Berkeley, CA. For those looking for further explanation of SIM, we refer you to John Meyer's AES paper "Equalization Using Voice and Music as the Source" (reprint #2150 I-8). — NP

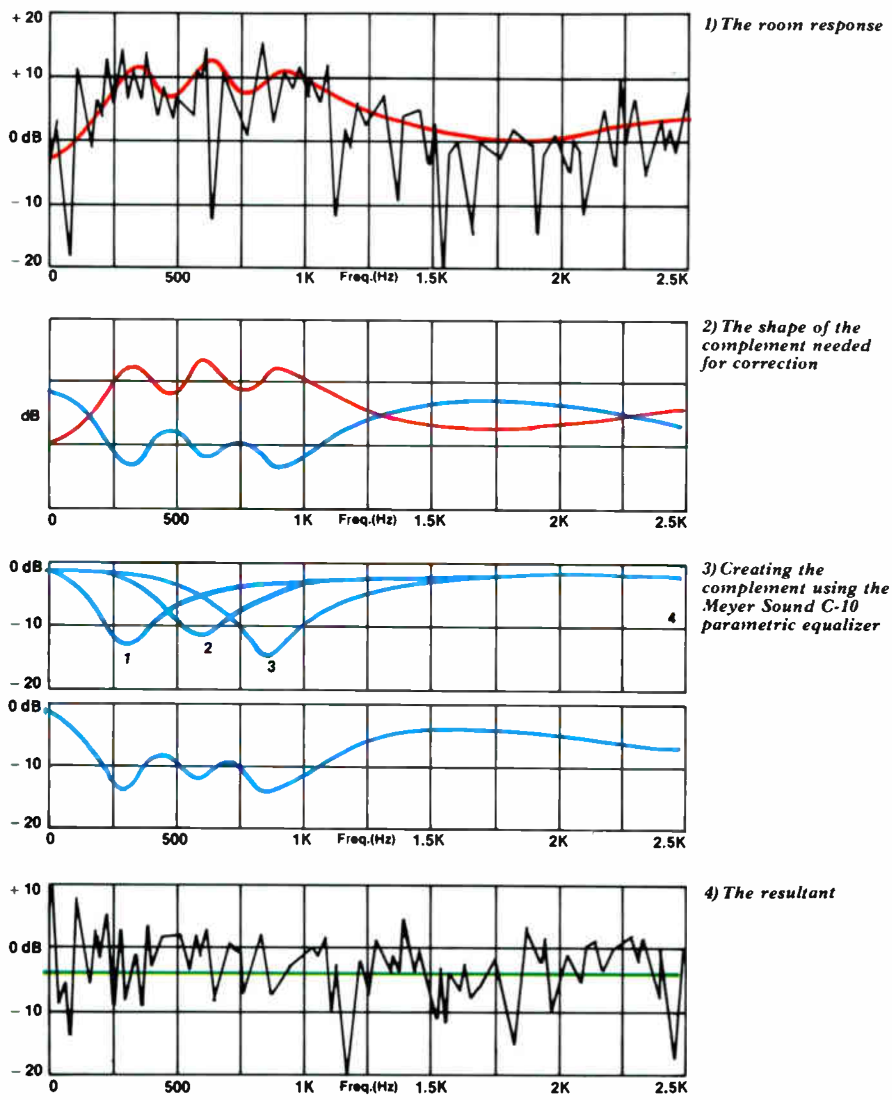


Figure 4: Correnting a Sound System with SIM

Chris Michie, a sound engineer, has worked on both recording and live sound for Pink Floyd, Roxy Music, and the Playboy Jazz Festival. At the time he wrote this article, Michie worked on in-house technical communications for Meyer Sound Labs. He now works as a staff writer for Hi-Tech Public Relations, San Francisco, CA.

NSCA Contractor's Expo

The 1986 National Sound and Communications Contractor's Expo, as predicted, had record attendence at its meeting in Las Vegas, NV, on April 28 to May 1. According to the NSCA, contractor attendees numbered over 1,000 with over 700 contracting companies represented.

"We are very pleased with these contractor attendance figures," Harold George, NSCA president, said. "These numbers show that electronic systems contractors have decided to pull together and support each other and our industry's suppliers, under the National Sound and Communications Association umbrella. We've seen NSCA and the Contractor's Expo grow dramatically in the last six years, and I'm proud to say that NSCA and the Contractor's Expo have become the major industry focus for the electronic systems contractor."

Harold Lander, chairman of the Board of the NSCA, said, "In a few short years, NSCA and the Contractor's Expo have grown from a perceived 'group of upstarts' to a highly respected force helping to steer the electronic contracting industry. The 1,000 plus registered NSCA contractors, plus the 300 or so contractors from EDS that spent time at our show, and the 3,100 industry people total in attendence at the Contractor's Expo '86 confirms this."

The Contractor's Expo '85 in Orlando drew 631 contractors. Mel Wierenga, NSCA vice president, said that he attributes the 40 percent increase in attendance to a variety of reasons. "The favorable reports from last year's show had a lot to do with the increases this year," said Wierenga. "Effective advance promotion and publicity from companies, and our friends in the trade press, also contributed. Our strong educational program drew contractors, because they are notoriously hungry for information at the technical and management levels. And, of course, the consistent quality of our exhibitors continually draws contractors."

Technical & Management Seminars

A major part of the Contractor's Expo each year are the technical and management seminars. According to the NSCA, attendance at the seminars this year was up significantly, which correlates with the overall increase in attendance. The NSCA reported that during each day of the seminars, over 500 people attended the technical seminars and over 200 attended the management seminars.

"Thanks to the expert faculty that addressed the seminars with great enthusiasm, the evaluations and responses we received were all positive," Bud Rebedeau, NSCA executive secretary, said. Topics covered at the seminars included *Current Research in Speech Intelligibility, Predicting Loudspeaker Performance by Inspection, Feedback—Where It Comes From, Where It Goes, Expanding the Sound and Communications Market, Fiber Optics, Local Area Networks, Digital Voice Technology*, and *Creating and Using a Clean Technical Ground.* The keynote speaker was Jack Berman of the Berman Institute of Agreeable Selling who gave a speech called *New Selling Techniques For Today's New Buyers.*

According to Mary Beth Warden, director of Member Services, the association will be polling its members as to what seminars they would like to attend at the 1987 Contractor's Expo.

New at the Expo

As with all trade shows, there were many exciting new products introduced at this year's Contractor's Expo, many of which we will be presenting to you in this and future issues of *Sound & Communications.*

Bose Corporation took advantage of the Expo to introduce its Modeler, a computer-aided design program for sound systems. The introduction was a major success for the company, according to John Stiernberg, field sales manager for Bose. "We at Bose decided that we would take advantage of the consistently excellent number of decision-making contractors the NSCA Contractor's Expo has provided for us over the years, and host 400 contractors at the introduction of our new product. NSCA Contractor's Expo '86 provided the perfect forum for Bose to accomplish this. The product was well received, the support from NSCA was great, and I would recommend this approach to other exhibitors," he said.

Yamaha Corporation debuted its line of microphones to the contracting industry. In the company's first venture into the microphone market, Yamaha introduced three vocal mics and two instrument mics. Two of the vocal mics and one of the instrument mics feature the industry's first microphone diaphragms made of beryllium.

Beyer Dynamic, which announced last fall that it would be introducing a new product line to the contractor industry, introduced products from that line at this year's Expo. The line includes a modular rack mount equalizer, a frequency shifter, powered mixers, and power amplifiers.

Numark's Professional Products Division, exhibiting for the first time at the Expo, announced the introduction of its CD9000 variable speed compact disc player. The unit incorporates many features never before available in a professional CD player such as speed and pitch controls, a remote control, and a special program-end indicator which alerts the user when a particular section has 30 seconds of playing time remaining.

Also new to this year's NSCA Expo was NSCA-TV News, a news and information program about the events and introductions at the NSCA Expo and its technical and management seminars. The news show, which was written, edited and produced by the editors and publisher of *Sound & Communications*, aired on televisions at the opening of the convention floor and in all the hotel rooms at the Sahara. Tapes (VHS and Beta) of the show will be made available to NSCA members through the services of the NSCA Video Library.

PHI TECH DIGI-VOICE FIRST TIME AT EXPO

Another new exhibitor at the Contractor's Expo was PHI TECH with its line of Digi-Voice products, microprocessor-controlled cassette decks, and MP-7 timer/controller. The Digi-Voice 1000, 256, and 32, all feature ADPCM (adaptive pulse code modulation) digitizing, ranging from single-channel access of up to 1,000 messages, to 32 seconds of audio on the lower-cost 32 model. Total record time is up to 2.77 hours, and the unit can be controlled from contact closure for remote start or RS-232. The Digi-Voice 1000 has the capability of "message synthesis" from a library of 1,000 digitized voice-recordings using a host computer.

The MP-7 is a microprocessor-controlled seven day timer/controller that can be programmed to start and stop up to eight different functions independently at preset times. Up to 500 different times can be programmed into the system, and they can be repeated on a daily or weekly basis.

Also on display was PHI TECH's Pro Com 60MP message repeater, Search 400 MP, and Voice Log II.

Circle 42 on Reader Rasponsa Card

ELECTRO-VOICE MT-4 CONCERT SPEAKER SYSTEM

E-V must have been working busily in their labs to come up with The Thunderbolt project. The fruits of this project were the refinement and implementation of an acoustical theory that E-V refers to as "manifold technology." Dave Carlson, project engineer, explained the concept to us, "...the combining of the acoustic outputs of a plurality of transducers into a device that has a single exit." The result is the capability to have the power handling of several drivers in the effective space of one, eliminating phase-shift and distortion problems of previous technologies.

The MT-4 consists of two boxes each 36 x 36 x 30-inches deep, the MTL-4 Low-Frequency Enclosure (loaded with four 18-inch woofers), and the MTH-4 High Frequency Enclosure (loaded with four 10-inch mid-bass drivers, four mid-range compression drivers and four high-frequency compression drivers). The full power output of this system is rated 134 dB at one meter, at very low distortion levels. While what we heard at the NSCA Expo was a "prototype," its sound quality could certainly be categorized as "hi-fi."

Circle 43 on Reader Response Card

BSM'S MODULA SWITCH MATRICES ALLOWS FLEXIBILITY

A new entry into the audio contracting market was the broadcast-electronics manufacturer BSM which showed its Modula routing switchers. With MIDI, musicians have the capability of "phoning in" their instrumental parts – now the audio guys have been given the opportunity to do the same with Modula Mini Modula routing switchers. BSM's routing switchers can be as simple as an 8x8 or as complex as a 256 x 256 (or even beyond with simple modifications). The Modula systems provide control over both video and audio signals, with a maximum of up to eight different levels that can be switched simultaneously. The entire network is controlled in a parallel 22-bit architecture providing fast switching response times. The processor will accept control information directly from remotes via party line coax, and/or from an IBM PC or RS-232 or via modem. The modularly expandable system has low power consumption, and it does not need fans. The lack of a fan requirement and comprehensive shielding ensure high signal integrity and purity. The remote control network features "Collision Detection System," which provides real-time programming and instant switch verification regardless of the number of remotes or the system size.

Circle 44 on Reader Response Card

OXMOOR REMOTE CONTROL FOR LEVEL AND RATE CHANGES

Yet more proof that digital electronics and robotics technology is trickling down to audio was in evidence at the Oxmoor booth. By using a shaft encoder that translates knob movements into a string of digital pulses, the RC-16 remote can precisely control changes in level, and rate of change. The RC-16 communicates its information on a four-wire daisy-chain buss, and uses RJ-11 modular phone plugs. Up to four-controls can be placed in a channel,

and up to 64 channels can be wired on a single control loop. LED's have replaced the pointer and all controls in a loop track each other. Options provide for priority, and key-switch lockout. The RC-16 controls the Oxmoor single rack-space DCA-2 Digital Control Attenuator, which has two-channels of balanced line in/out with 29 steps of 1.5 dB per-step attenuation plus a 90 dB off position. The maximum cable length to the farthest remote is 2,000 feet (based on typical modular telephone cable's resistance of 12 ohms per 330 feet). With other features that include optional output transformers, front-panel gain controls, remotes that are ready-to-mount in single or double gang electrical boxes and preset for priority level upon external contact closure, the RC-16/DCA-2 promises to eliminate dirty pots.

Also on display, was Oxmoor's new 4x4 buffer amplifier with inputs that may be assigned to any of four mixers (which is followed by a ±20 dB gain stage), and any of the four mixers may be assigned to any of the four balanced output amplifiers— this single rack-space unit will bale us out of having to build little black boxes.

Circle 45 on Reader Response Card

HARRIS/MOTOROLA DEBUTS P-TEC/5 CEILING LOUDSPEAKER

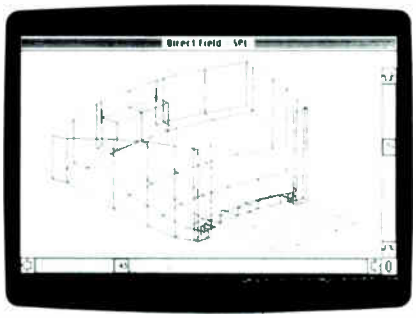
Harris/Motorola has managed to reintroduce piezo-electric technology as implemented with ceramics in its new P-TEC/5 ceiling loudspeakers. Harris/Motorola's newly developed process of slicing a piezo-ceramic material to a mere

.004-inch thickness affords several advantages over electromagnetic drive mechanisms in loudspeaker designs: the speaker's total weight is less than one-half-pound, 25/70-volt line-transformers are eliminated, and only a fraction of the power is required to drive the loudspeakers. Besides the simple time/cost savings on installation of the loudspeaker; since it is surface mounted, no back-box is required. At the Harris/Motorola booth they demonstrated the voice-range loudspeaker's immunity to moisture by dunking it into a fish tank complete with goldfish.

Circle 46 on Reader Response Card

BOSE INTRODUCES MODELER SOUND SYSTEM SOFTWARE

Bose has made a new commitment to the sound contracting world with the introduction of the Sound System Software family. The demonstration of the first in the series, Modeler Design Program, took place in the Riviera Hotel's La Cage Showroom. The loudspeaker system in the showroom was designed by Bose with the Modeler software. Thus, the presentation allowed for a lot of ground to be covered in a short time—watching the screen display of the system's design and being able to look around the room where the system was installed really drove the point home.



Taking advantage of the enhanced graphics of the Apple Macintosh, Modeler is based upon representing the sound system's environment, which can be an enclosed room, or an outdoor facility, as a series of adjoining planar regions. The program enables the user to rapidly construct complex spaces with up to 100 planes and 100 loudspeaker clusters. Also, the three-dimensional model of the room may be rotated in any direction. Analysis of sound system performance includes calculation of direct-field SPL, time-arrivals from individual sources, and Sabine reverberation time. Among some of the features of the program are: the depiction of a sound system's coverage with grey-scaling (numbers are available too) where "hot-spots" are bright, and "dead-spots" are dark; the ability to view a loudspeaker's sound output as a three-dimensional "directivity balloon"; and a library of 12 standard materials and their absorption coefficients in octave-bands, with the ability to add as many materials to the file. The "open" data-base structure of Modeler will easily allow the input of any loudspeaker's data to be input. In fact, Modeler comes with JBL's data, and we were told that Community will follow shortly.

Circle 57 on Reader Response Card

CRESTRON MICROPROCESSOR—BASED CONTROL SYSTEMS

New to the NSCA Expo, Crestron introduced to contractors a comprehensive line of microprocessor-based remote-control systems. Crestron demonstrated its Executive Series II, Crestnet, and Crestlite systems.

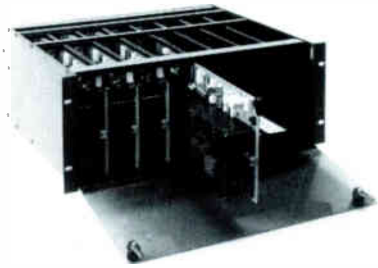
The Executive Series II and the Crestnet are capable of remotely controlling any piece of equipment that will accommodate external control, and with their interface modules virtually any AV equipment, motor, or AC power. Remote panel controls are software assignable, and any panel can be installed up to 1,000 feet from the master unit.

Crestlite, a dimming system for incandescent and flourescent lighting fixtures, provides for seven zones of control with up to seven programmed preset levels. A feature of Crestron's systems is the ease with which UHF wireless remotes are incorporated into the system. Handheld remotes can control up to 16 functions, while wireless control-panels can control up to 48 functions. The Crestnet, which is based around the CMU-1 Master Unit, uses LAN (local area network) technology and all remotes can be daisy-chained via four-wires. Also, the CMU-1 may be entirely programmed from a host PC and various "setups" may be saved on disks for future use.

Circle 48 on Reader Response Card

IED'S NEW 4000 SERIES AUTOMATIC MIXER

IED showed its new 4000 Series Automatic Mixer, which is the only such system that can be provided with automatic programmable gain controls on each input. This system can be operated by IED's new 590, MS-DOS-based computer, which allows for complete setup and change from the combining of many rooms to individual function setups. Other features of the system are: each input and output circuit can be fitted with applications-dictated audio processing options; the 4422R remote-control card that allows for manually reconfiguring room setups, without a computer or outboard switching matrix; and when combining rooms, all active busses are combined.



IED also showed its 6270 switching-amplifier. The 6270 is rated at 200 watts which directly drives a balanced 70-volt line. The amplifier which operates 90 percent efficient, is a card which is housed in a four-space main-frame that can accommodate up to eight amplifier cards—1,600 watts in seven-inches!

Circle 49 on Reader Response Card

TECHPRO'S MS731 INTEGRATES TALK BACK SPEAKER STATION

Technical Projects has introduced the MS 731, a two-channel master station that integrates a talkback loudspeaker station. The new master station operates in full duplex mode with a plug-in gooseneck microphone, in push-to-talk mode using the built-in speaker, or with a headset. The 731's override generator shunts loudspeaker stations into their override mode. Miniature switches provide routing for the auxiliary input signal and allow programming of the override function.

Technical Projects has also introduced the BP-112 and BP-113 two-channel belt packs. These portable headset stations allow the operator to speak and listen over one or two channels. Each channel has its own mic switch, on/off switch, earphone, volume control, signal switch, and signal light. The BP-112 has a pair of three-pin XLRs, each for connection to one communications circuit. The BP-113 features dual six-pin XLRs to permit loop-through of two circuits.

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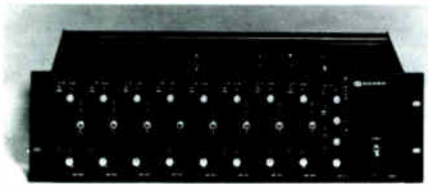


BOGEN AUTO MIXER MODULE WITH DYNAFEX

The latest introduction to Bogen's HI-TEK Series is the MM-SM6 automatic microphone mixer module. The mixer may be wired with two levels of priority and features Dynafex noise reduction. Additionally, the mixer may also incorporate Bogen's MM—S02A three-band EQ modules. The module, according to Bogen, is particularly suited for use in courtrooms, churches, auditoriums, boardrooms, and other areas requiring multiple microphones with varying levels for each. The specs for the MM-SMG are frequency response 20 Hz to 20 kHz, THD .05 percent or less, typical priority attenuation -30 dB, and maximum gain of 45 dB nominal.

Circle 51 on Reader Response Card

CETEC VEGA DEBUTS PRO 1 WIRELESS MIC SYSTEM

Cetec Vega has introduced its PRO 1 professional wireless microphone system in two versions—the PRO 1-B bodypack and the PRO 1-H hand-held system.

The PRO 1-B system consists of the T-37 bodypack transmitter and R-31A receiver. The PRO 1-H system consists of the T-36 hand-held transmitter and R-31A receiver.

The R-31A PRO receiver features two LED bar graph displays—one for RF signal level and the other for audio level. The preselector is a two-pole helical-resonator filter, silver-plated for low loss and long-term durability. The power transformer is a high-performance toroidal design for low hum; both 115- and 230-VAC operation is supported, as is external DC operation.

Circle 52 on Reader Response Card

TEKTONE INTRODUCES TELEPHONE ENTRY SYSTEMS

TekTone Sound & Signal has introduced the Tek-Entry telephone entry systems. The TE series is a complete, automatic call and entry control system. The TE units operate on a standard telephone jack and 110 VAC and are available with a telephone handset or a hands-free speaker/microphone.

Models are available with a number capability of from one to 1,000 units and feature vandal-resistant, modular aluminum construction, front panel programming, crystal decoder for guaranteed correct touch-tone decoding, memory retention during a power failure, and selectable one or four minute talk time limit.

Circle 53 on Reader Response Card

ESG INTRODUCES OUTDOOR SPEAKER SYSTEM

Electronic Systems Group has introduced its new SoundScape outdoor speaker systems. Typical applications for the weather-resistant speaker system include pool and lakeside recreation areas, hotel atrium and lobby gardens, theme parks, zoological gardens, parks, and residential gardens and patios.

The ATS-360 SoundScape ground-speaker Series employs an inverted, 360–degree direct-radiating speaker. The speaker is said to eliminate trapped-water problems. An integral metal screen basket/cone shield also acts as an effective pest barrier. There are no internal reflectors required in the system.

The low-profile ATS-360 Series cases are made from polyethylene and include impregnated ultra-violet ray inhibitors to resist color fading. The cases are built to withstand the normal abuses of landscape trimmers and gardening tools.

Circle 54 on Reader Response Card

NEW SOLID STATE INTERCOM FROM DELTACOM

Deltacom has introduced the 2600 Series intercommunication systems. The systems feature solid state design, modular construction, hybrid output stages, toroidial transformers, two-channel telephone option, and selective time-tone option. Other features include 'Color-Flo' operation instructions on the control panel, dual-channel intercom/program capability, 120-watt program amplifier, 10-watt intercom amplifier, bar LED volume indicator, one-button general announce with program auto-mute, choice of privacy options, pre-announcement chime, and complete factory assembly.

Circle 55 on Reader Response Card

DIGITAL MESSAGE REPEATER RECORDS/PLAYS TO 40 MINUTES

MacKenzie Labs has introduced the DMR/RP Digital Message Repeater Record/Play unit. The (DMR/RP) is all solid-state and has no moving parts. Messages are recorded on-the-spot by talking into a microphone or via any line level source such as a tape or cassette recorder. The audio is automatically converted into digital information and stored in memory chips (RAM). Messages remain in RAM until erased overtly by recording a new message. Upon command, messages are converted from digital to analog and played instantly via the unit's audio output.

The DMR/RP can be furnished with message lengths from 40 seconds to 40 minutes.

Circle 56 on Reader Response Card

TAPE-ATHON DEBUTS TWO COMPACT POWER AMPS

Tape-Athon has introduced two new compact power amplifiers, the 10-watt AM251A and the 20-watt AM252A. The solid state amplifiers feature two-channel mixing and four- or eight-ohm output impedance. The front panel features a "music" control, a mic control, a tone control, a power switch, and a pilot light. The amps are rack mountable in a one-half-unit space or desk mountable. The rear panel features a phono "music" input jack and a screw-terminal microphone input. Both units feature frequency response of 50 to 20,000 Hz with less than two percent THD, according to Tape-Athon.

Circle 58 on Reader Response Card

PRODUCTS IN REVIEW

a closer look

by gary d. davis

BOULDER 500 POWER AMP

Boulder Ampilifiers has introduced a power amplifier, the Model 500. The unit utilizes discrete 990 Operational Amplifier technology, originally developed by Deane Jensen, relying upon DC servo feedback.

There are two separate stages. First, an 18 dB differential input gain stage. Second, a modified version of the 990, which includes the 14 output transistors in the feedback loop with a closed loop gain of only 2.5 (8 dB). This translates to over 100 dB of feedback around the output transistors, bringing the distortion down to .005 percent at 20 kHz at full power, and .0015 percent below 2 kHz.

This approach is said to ensure low distortion, low noise, high slew rate, and gain-bandwidth products beyond the capabilities of IC opamps. The design was achieved using extensive computer optimization of circuit values, as well as computer matched components held to within .01 percent tolerance for extremely high common mode rejection (80 dB at 60 Hz, 70 dB at 10 kHz).

The very conservative, 20 kHz bandwidth power ratings are 150 watts into 8 ohms (per channel, stereo mode), 250 watts into 4 ohms, or bridged mono 500 watts into 8 ohms. In fact, there are 28 metal-case 250-watt rated output transistors for a total of 7,000 watts rated capacity, ensuring an ample SOA, with a damping factor of 800 at frequencies below 1 kHz, and 100 at 20 kHz.

Harmonic distortion at rated power, from 20 Hz to 2 kHz is no more than .0015 percent, and rises to no more than .005 percent full power at 20 kHz. The closed loop, maximum voltage slew rate is 35 V/uS (stereo) and 70 V/uS (mono). With a 100 kHz power bandwidth, rated frequency response is 20 Hz to 20 kHz, +0.00, -0.04 dB. (Small signal bandwidth is 200 kHz.) S/N ratio is 115 dB, 20 Hz to 20 kHz, unweighted, and convection cooling avoids fan noise. Gold-plated input connectors include XLRs, phone jacks and phono jacks; impedance is 10 K ohms balanced, 25 K ohms unbalanced. Output is via gold-plated five-way binding posts.

The unit utilizes a 1,200 watt toroidal power transformer, with four filter caps totalling 148,000 uF so there is ample reserve for bass notes. Low flux leakage allows use near tape machines or turntables. The solid one-eighth-inch aluminum chassis is heavily shielded (including power cord) to operate in high RF fields. All six PCBs are fiberglass epoxy for maximum durability. The unit weighs 51 pounds and occupies seven inches in a standard 19-inch rack (or is free standing).

The conductive plastic level controls are recessed, as is the power switch. LED's are provided for each channel for input offset, output short, thermal warning (pre-protect) and thermal protect, as are level LEDs. According to Boulder, the amp is basically overload proof. In the Boulder 500, both voltage and current are measured and linearly compared to the operating temperature to determine if limiting is necessary. This technique permits high, continuous maximum power and even higher power (more than 50 amps) for short term transients. A comparator approach means the protection circuitry is completely off before the current limiting threshold is reached, which avoids distortion induced by the protection circuit. Short circuits are detected by monitoring current clipping at low voltages (the circuit cycles off for three seconds and tries again). To prevent DC amplification, the output will not turn on unless there is less than 100 mV DC at the input.

Comments: The Boulder 500 is an example of how computer circuit optimization can be applied to feedback stabilization, the goal being sonic clarity.

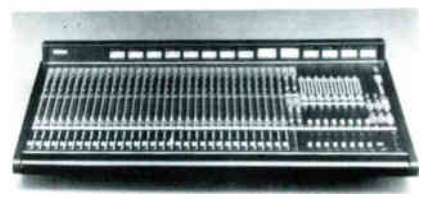
Capacitors are a major source of potential audio distortion. The use of Jensen 990 operational amplifiers with DC servo feedback eliminates all coupling capacitors in the audio circuit.

It is widely thought, especially by hi-fi people, that too much feedback sounds bad. These people are concerned with the transient distortion which results from the feedback arriving late due to the delay in the large geometry transistors.

According to Deane Jensen, Boulder has solved that problem using Jensen's COMTRAN circuit analysis and optimization program on a Hewlett-Packard computer. They optimized the values of the components which affect the open loop response to minimize the transient distortion at the fixed closed loop gain of 2.5. This resulted in an amplifier with no overshoot at all, even with a reactive load (1.5 ohms at 60 degrees impedance), with 200 kHz bandwidth closed loop. This has never been done before.

At $2,650, the Boulder 500 certainly is not for everyone. However, its unusual design, claimed superior sonic quality, unconditional stability, and 100 percent computer-matched and tested construction should warrant your closer look.

Circle 59 on Reader Response Card



YAMAHA® PM3000 AUDIO MIXING CONSOLE

The new Yamaha PM3000 audio mixing console has a wide range of features useful for theatrical sound reinforcement.

The console's eight Master Mute groups, together with the eight Mute assign switches on each input module enable all the sound sources for a given scene to be preset so they can be turned on or off.

The console has up to 94 dB of gain so that distant microphones and quiet speaking voices pose no problems, according to Yamaha. When less amplification is needed, the PM3000's eight VCA groups make it possible to alter the balance of different groups of inputs. The VCAs can affect all outputs from an input module, and they can control overlapping groups of inputs for "additive" or "subtractive" fades.

The console's Mix Matrix can be used as an assignable output mixer. Similar to a lighting console in concept, the Mix Matrix permits up to 11 sources (the eight group busses, the stereo bus, and matrix sub inputs) to be remixed into eight different output mixes. The matrix outputs can drive various primary speaker systems, effects speaker systems, as well as lobby, dressing room, and other remote speakers. The inputs to the matrix can be mixed independently, as required, for each of the areas.

If a simultaneous recording is needed, the matrix can be set to mix signals from ahead of the group and stereo master faders, so the group and stereo outputs can be used for independent multitrack and

(continued on page 50)

DATAFILE *info. sources/new literature*

Interconnect Pocket Reference Guide From Samtec

Samtec's new *Interconnect Pocket Reference Guide* includes sections on pin out configurations, contact and termination design, cable connectors, materials, applications, quality standards, and conversion tables. Discussions include comparisons of machined and stamped contacts, and the performance advantages of each system. The guide, which is free, was designed for new technical employees and experienced engineers.

Circle 33 on Reader Response Card

Bi-Tronics Catalog Includes Connector Location

Bi-Tronic's new 1986 catalog includes the Connector Locator Service. The service is said to eliminate problems installers have locating essential foreign or domestic made connectors, identifying a connector used for repair or replacement on a foreign or domestic product, or selecting the connector for a specific application.

Circle 34 on Reader Response Card

NATA Telecommunications Sourcebook Now Available

The North American Telecommunications Association has released its *1986 NATA Telecommunications Sourcebook*. The *Sourcebook* covers new equipment, manufacturers, contractors, suppliers, services, and public policy developments.

In the *Sourcebook*, industry participants will learn about critical marketing issues such as the Centrex revival, privately owned pay phones, data peripheral and other new markets, and private network evaluation.

Circle 35 on Reader Response Card

Carter-Craft Updates Video-Cable TV Workbook

Carter-Craft has released its new *Video-Cable TV Workbook II*, an updated, 44-page version of the original *Video-Cable TV Workbook*. The new *Workbook* features step-by-step guidelines and illustrations for hooking up TVs, stereo-simulated TVs, video cassette recorders, and video disc players. The *Workbook* also provides tips on dubbing, cameras and camcorders, and preventive maintenance.

Circle 36 on Reader Response Card

Remote Module Guide Published by FSR

A new application brochure from FSR covers remote control modules suitable for sound or video installations. Developed for contractors and consultants, the brochure covers topics on remote volume, power switching, mic switching, slide and tape machine control, VCR control and video switchers.

Circle 38 on Reader Response Card

AD INDEX

SOUND & COMMUNICATIONS

QUICK PRODUCT INFORMATION GUIDE

You can quickly and easily obtain additional information about products and services featured in this issue of **Sound & Communications** by completing this card now.

SOUND & COMMUNICATIONS

JUNE 1986

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☐ G Advertising/Promotion
☐ H Other ______________

3. **Primary business of company (only one):**
☐ 1 Contractor—Engineered Sound/Acoustical
☐ 2 Contractor—Interconnect/Intercom
☐ 3 Contractor—Fire/Alarm/Safety
☐ 4 Electrical Contractor
☐ 5 Pro Audio/Studio/Reinforcement
☐ 6 Architect/Designer
☐ 7 Engineering/Acoustical Consulting
☐ 8 Maintenance/Service
☐ 9 Dealer/Distributor/Rep
☐ M Manufacturer
☐ 0 Other ______________

4. **Your purchasing authority is:**
☐ Final approval/Buyer
☐ Recommend/Specifier
☐ No Direct Authority/User

5. **Intensity of your product need:**
☐ Have salesman call
☐ Need within 3-6 months
☐ Future projects

6. **Number of employees at your company:**
☐ 1-3 ☐ 4-10 ☐ 11-25 ☐ 26-100 ☐ over 100

EXPIRES 10/1/86

NEW PRODUCTS: (Please Circle)

			43	60	77
			44	61	78
			45	62	79
1	15	29	46	63	80
2	16	30	47	64	81
3	17	31	48	65	82
4	18	32	49	66	83
5	19	33	59	67	84
6	20	34	51	68	85
7	21	35	52	69	86
8	22	36	53	70	87
9	23	37	54	71	88
10	24	38	55	72	89
11	25	39	56	73	90
12	26	40	57	74	91
13	27	41	58	75	92
14	28	42	59	76	93

ADVERTISED PRODUCTS: (Please Circle)

			243	260	277
			244	261	278
			245	262	279
201	215	229	246	263	280
202	216	230	247	264	281
203	217	231	248	265	282
204	218	232	249	266	283
205	219	233	250	267	284
206	220	234	251	268	285
207	221	235	252	269	286
208	222	236	253	270	287
209	223	237	254	271	288
210	224	238	255	272	289
211	225	239	256	273	290
212	226	240	257	274	291
213	227	241	258	275	292
214	228	242	259	276	293

SOUND & COMMUNICATIONS

JUNE 1986

QUICK PRODUCT INFORMATION CARD

NAME: ______________________ TITLE: ______________________

COMPANY: ______________________

STREET: ______________________

CITY: ______________________

STATE: ______________________ ZIP: ______________________

TELEPHONE: ( ) ______________________

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☐ E Field Installation/Maintenance
☐ F Consultant
☐ G Advertising/Promotion
☐ H Other ______________

3. **Primary business of company (only one):**
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☐ 2 Contractor—Interconnect/Intercom
☐ 3 Contractor—Fire/Alarm/Safety
☐ 4 Electrical Contractor
☐ 5 Pro Audio/Studio/Reinforcement
☐ 6 Architect/Designer
☐ 7 Engineering/Acoustical Consulting
☐ 8 Maintenance/Service
☐ 9 Dealer/Distributor/Rep
☐ M Manufacturer
☐ 0 Other ______________

4. **Your purchasing authority is:**
☐ Final approval/Buyer
☐ Recommend/Specifier
☐ No Direct Authority/User

5. **Intensity of your product need:**
☐ Have salesman call
☐ Need within 3-6 months
☐ Future projects

6. **Number of employees at your company:**
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5	19	33	59	67	84
6	20	34	51	68	85
7	21	35	52	69	86
8	22	36	53	70	87
9	23	37	54	71	88
10	24	38	55	72	89
11	25	39	56	73	90
12	26	40	57	74	91
13	27	41	58	75	92
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210	224	238	255	272	289
211	225	239	256	273	290
212	226	240	257	274	291
213	227	241	258	275	292
214	228	242	259	276	293

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BOOK REVIEW

by Ted Uzzle

Basics for System Operators

Everest, F. Alton, *Successful Sound System Operation*, Blue Ridge Summit, PA, TAB Books, 1985, xii, 321 pp., $17.95 (paperback)

Most books about audio and acoustics try to do too much. Perhaps the publisher thinks book-buying audio people are few and far between so each book must have a little something for everyone. The result is that our bookshelves groan with 20 copies of the same book. Twenty different authors, 20 different publishers, 20 different titles, but *the same book*!

TAB Books is an exception. They have a wide selection of audio books, and their books tend to be aimed at quite specific audiences. A number of TAB Books on audio have been written by F. Alton Everest. He writes especially useful ones because he targets them to specific types of readers, and then assembles material from the universe of audio information that will be directly, practically usable by those readers.

This book, *Successful Sound System Operation*, is for sound system operators. It is not for designers (there are other books on that), it is not for installers (again, there are other books), and it is not for salesmen (we haven't seen any books on sound system sales). It is suited to operators, and has the information they need to do their jobs better.

Successful Sound System Operation will probably teach the experienced contractor or consultant very little. Yet, the contractor and consultant may well find the book vastly useful to present to system operators in those installations where he suspects the operator doesn't know as much as he should.

Remember that the entry-level job for sound system work of all types is system operation. This job is also the neck of the bottle from which flows customer satisfaction with the system. A properly designed and installed system that is badly operated makes customers unhappy, and an operator who doesn't know any better (or who has an agenda of his own) may blame his problems on the designer, the installer, or the manufacturer. The rest of the system team can no longer afford to be indifferent about the quality of operation.

There's an opinion held in some quarters that the operator should be kept ignorant, lest he begins tinkering with the system and making a constant stream of small, undocumented changes to it. These people tend to quote the poet: "A little learning is a dang'rous thing/Drink deep, or taste not the Pierian spring..."

I am familiar with one prominent installation where the contractor and the consultant demanded, and received, the right to make hiring recommendations on operators! Each interview began with the question, "Do you know anything about sound system design? If the hapless applicant answered "yes," he was asked to step through a door...and found himself out in the parking lot!

More mature reflection would suggest that whether or not an operator tinkers has almost nothing to do with how much or how little he knows. It has to do with how carefully his authority is defined to him, and the level of discipline his employers exert over him. It has to do with how much confidence he has. And that if there's a problem he can get help or advice from the

(continued on page 49)

FACES AND PLACES

JOEL E. KAHN

Bogen Announces Appointments Of Kahn, Ramsaran, Spencer

Bogen has announced three appointments, Joel E. Kahn as director of marketing, Ashook Ramsaran as director of engineering, and Walter J. Spencer as product manager for engineered systems.

Kahn comes to Bogen after 14 years at Executone, where he rose from sales administrator to director of marketing. Kahn graduated from Queens College with a degree in business administration and economics.

Ramsaran, who holds a BSEE and a MSEE from the Polytechnic Institute in Brooklyn, NY, previously served as director of engineering at Executone.

ASHOOK RAMSARAN

WALTER J. SPENCER

Spencer comes to Bogen from the contracting firm of Sound Plus, where he was general manager.

PHILIP J. LANTRY

TOM MILAN

Audio-Technica Announces Personnel Changes

Audio-Technica has appointed Philip J. Lantry as regional sales manager, professional products. Lantry has begun working in Audio-Technica's Stow, OH, headquarters, coordinating the efforts of the firm's national sales reps.

Before joining Audio-Technica, Lantry worked in a sales position with a realty firm while pursuing a second career as a performing and recording musician.

In other news, a promotion and an appointment have been made in the marketing department of Design Acoustics, the loudspeaker systems manufacturing division of Audio-Technica U.S., according to Ken Reichel, the firm's vice president of marketing. Promoted to marketing manager of the division is its former product specialist, Tom Milan. Newly appointed as product specialist is Gary S. Post.

BOOK REVIEW

(continued from page 47)

consultant or the contractor. It has to do with the presence of small problems in the system that seem never to be cured completely. It has to do with demands made on him and the system for which it was not designed. If he's not going to tinker, it doesn't matter if he is very knowledgeable. If he is going to tinker, the greater his knowledge the smaller the disaster he may make.

Where there may be a problem with operator quality, making a gift of *Successful Sound System Operation* to the operator serves many useful purposes. For one, it establishes the contractor or the consultant as a source of information. It demonstrates his concern that the system be operated well. It predisposes the operator to work with the sound system team if problems crop up later, rather than blaming them for the problems. It promotes the sale of consumable supplies (such as microphones), rather than tempting the operator to go out to Radio Shack and buy them himself. It puts the maker of this modest gesture first in line if the system later needs modifications or enlargements because of evolving use. It will also teach the operator quite a bit about sound.

Consider this: in those cases where the consultant and the contractor are worried about being bad-mouthed by the operator, it's virtually certain the operator is worried about being bad-mouthed by them. That's a vicious circle that spirals down, not up. The author touches directly on the relationship of these members of the sound system team in a brief section entitled, "The Consultant and The Sound Contractor:"

> *...know-how is required far above that of the average resident electronic wizard. Satisfactory sound seldom results from a budget sound reinforcement system installed by a well-intentioned, but inexperienced person.*

The book begins with introductory chapters on sound waves, on elementary electricity and electronics, the perception of sound, and what the author calls "that demon the decibel." Toward the end of the book there is a chapter covering the rudiments of reverberation and room dimensions.

The following chapters are on types of reinforcement systems, and the strengths and limitations of each, on microphones and their use, and on the safe use of loudspeakers. A later chapter deals with the selection and use of foldback loudspeakers, which are virtually always set up by the system operator. These chapters focus on the theoretical reasons some arrangements work and others don't rather than the more usual sets of tips and hints assembled from trial and error.

The chapter on the mixing console describes several sample models, including mix-power amplifiers and mixer-recorders. This is one of the longer chapters in the book, as it properly should be, since the operator will spend most of his time at the console. Yet, this chapter could be more useful if it had shown the way consoles must be customized for each different setup. Everytime there's a change in the microphone arrangement, everytime there's a change in the foldback arrangement, the connections to the console must be changed.

The book concludes with a group of chapters each describing one of the electronic boxes apt to find their way into modern sound systems: power amplifiers reverberation devices, limiters, compressors, noise gates, equalizers, signal delays, magnetic tape recorders, and automatic microphone mixers.

There is a chapter on lecterns with built-in amplifiers and loudspeakers. As much as professionals may groan at the idea of such devices, the fact is many of them are in use, and it is the operator who is expected to make them work.

Throughout *Successful Sound System Operation* the author has kept it aimed at, and useful to, the system operator. No other book known that I know of will be so quick about capturing the attention and rewarding the study of the beginning sound system operator, and many are sure to find it an introduction and vade-mecum for their first job in professional audio. Twenty years from now many professionals at all levels of our business will look back on this as the first book they read about sound, and the one that opened their way to many more articles and books.

A CLOSER LOOK

(continued from page 45)

two-track tape recording mixes. Control room outputs make it possible to monitor the console outputs while working in an isolated booth; they even carry the cue signals so that the operator doesn't have to wear headphones. A communication input and talkback output facilitate interface to intercom systems.

Comments: Yamaha International first unveiled its PM3000 series of mixing consoles at the November 1985 AES Convention. The PM3000 was also shown recently at the NSCA Expo in Las Vegas, NV. The line offers three models, differentiated by the number of input channels—24 input, 32 input, and 40 input. The 40-input model has its master controls located in the center of the board, easily enabling two soundmixers to work together while handling the large number of inputs.

Despite many design advancements incorporated into the PM3000, Yamaha has been very careful to keep the basic philosophy of its console line intact. Continuity of signal flow and routing remains constant, as does the color coding of controls (EQ, Aux, etc.). On short notice, most engineers already familiar with other Yamaha consoles will be able to "fly" this new board with minimum preparation. They can "ignore" most of the advancements until there is time to sit down and learn them.

According to Craig Olsen, professional audio products manager for Yamaha, the technology for the PM-3000 was developed "from over two years of research and evaluation. We spoke to tour engineers, mixers doing legitimate theater, recording consultants, sound contractors . . . everybody's input was considered."

The PM3000 series of consoles is geared to a broad sound reinforcement field, stressing the fixed installation and theatrical sound sectors. The boards are electronically balanced (inputs and outputs), although input and output transformers are available as options.

Flexibility is the main emphasis in the design of the PM3000. Its muting system provides eight master mute groups as well as eight mute assign switches on each input channel. This configuration allows any single input to be assigned to any specific, or combination of, master mute groups. This system of muting is especially good for theatrical and music work where scenes and song arrangements require certain mics to be infrequently used. Assigning specific inputs to a mute group enables the soundmixer to hit one switch and turn off whichever input has been assigned to that group. No unused microphones need remain open. Also on each input module is a mute safe switch (again in conjunction with the mute groups). The mute-safe switch can be set to prevent an always-open channel from being inadvertently muted.

Each input fader controls a VCA and is assignable to any or all of the eight VCA masters. Each VCA master raises and lowers the level of all inputs assigned to it. The VCA output of each fader is routed to the respective assigned subgroups.

Further evidence of the PM3000's human engineering and flexibility is the console's Mix Matrix, which, Craig Olsen said, can be used "as an assignable output mixer. It's an incredible tool. It permits you to provide different mixes to any source you can think of, whether remote recording trucks, different speaker feeds for dressing room cues, main speakers and effects speakers system. . . .

"Many people judge a console by how many busses it has, or by its number of assign switches. Technically, the PM3000 is an eight-buss console, that is, there are eight main subgroups. Nonetheless, it is possible to get 26 different mixes (eight matrix outputs, eight group outputs, stereo outputs and auxiliary sends) out of the console. There are jumpers inside each module that are all on switches to facilitate change. If you change the preset switches on the group master faders, for example, you could change the pickoff point to the matrix to pre-fader so you could do eight independent level mixes out of each of the matrix outputs, as well as all eight of the group outputs, because the signal going to the matrix is before the fader. At that point, the group fader no longer affects it. Continuing in this manner you can see how you can get 26 mixes out of the console. And all of this can be done without any soldering. Just pull out the module and you'll see all the switches clearly labelled. It's one of the means by which the board can be prepared for a number of different uses," he said.

The three PM3000 models are priced (suggested retail) as follows: 24 input, \$24,500; 32 input, \$29,500; 40 input, \$34,500.

—*H.G. La Torre*

La Torre is Technical Editor of *Music & Sound Output* magazine.

IMPULSE TESTING
(continued from page 27)

computation, without ever once banging the room with an N wave. Of course, some stimulus must be used and these include swept tones, psuedorandom noise sequences and natural sounds such as music itself.

I deliberately considered only the simplest method here, in the hope of stripping the "impulse response" of some of its mystery; of illustrating the importance of what it is telling us; and perhaps with the feeling that "you have to learn how to walk before you can run."

NOTES

1 L. Beranek, *Music, Acoustics & Architecture* (New York: John Wiley & Sons, 1962), pp. 63-71.

2 M. Rettinger, "The Statistics of Delayed Reflections," *JAES*, 16 (1968), pp. 436-439.

3 R. Heyser, "Acoustical Measurements by Time-Delay Spectrometry," *JAES*, 15 (1967), pp. 370-382.

4 M. Schroeder, "Integrated-Impulse Method Measuring Sound Decay Without Using Impulses," *JASA*, 66 (1979).

5 A. Berkhout et al., "Acoustic Impulse Response Measurement: A New Technique," *JAES*, 32 (1984), pp. 740-746.

Bill Lobb is an independent audio consultant whose major client is Jaffe Acoustics. Lobb also consults for Future Sound, Bozak, Ringling Brothers Circus, and other firms. Lobb is a member of the Audio Engineering Society.

CONSULTANT
(continued from page 11)

project, this will often involve late night hours when sufficient quiet is available for these procedures.

(9) Complete the project. At the conclusion of a project, sound contractors are frequently (and logically) tempted to move on to other new and profitable projects before completing the details of the project at hand. Final punch list items, system documentation, and as-built drawings are the usual victims of such temptation. As such, final payment of 10 to 25 percent of the contractor's value should be withheld until it is determined that the specification workscope has been completed in every respect.

(10) The contractor should observe carefully the specified terms of the warranty and provide assistance to the user during the first few weeks of operation. A proposal for a preventative maintenance contract should be submitted by the contractor for services beyond the expiration of the warranty period.

These concepts are most applicable to larger installation, but are frequently appropriate for smaller sound systems as well. They are certainly not the only means of improving the process of construction of sound systems.

Marc Beningson is a senior electro-acoustic consultant with Jaffe Acoustics. Beningson, a member of the Audio Engineering Society, came to Jaffe Acoustics shortly after receiving a B.S. in Mechanical Engineering from the Rensselaer Polytechnic Institute.

CORRECTION

In the May issue of *Sound & Communications* on page 24 the headlines on the pie charts are reversed. The headline on the chart in the first column should read "Permanent Employees." The headline on the chart in the thrid column should read "Gross Income."

SALES & MARKETING

(continued from page 10)

to look at the other benefits of teleconferencing that are more difficult to quantify:

(A) Increases Productivity and Efficiency
- a. Reduce unproductive time
- b. Prevent meeting delays
- c. Hold shorter and more structured meetings
- d. Optimize meeting attendance

(B) Improve Management Communications
- a. More interface at all levels
- b. Increased flexibility

(C) Enhanced Business Opportunities
- a. Customer service
- b. Competitive advantage
- c. Access to information
- d. Immediate information exchange
- e. Access to experts
- f. Time-share scarce talent
- g. Quicker decisions.

Value of Increased Efficiency: One way of looking at the "soft" benefits is to consider their impact on the profit of the corporation. By squeezing more time into a day, through more efficient communications, the number of available hours per manager will increase. Teleconferencing can be used to help managers make better use of their time, and thus accomplish more in a shorter time span. The greater the number of productive hours available to each manager, the higher the management profit and thus the corporate profit. If teleconferencing is utilized to squeeze more hours into the day, the greater the usage of teleconferencing, the more hours available.

Needs Assessment Applications

Understanding the methodology and rationale behind a needs assessment is only one part of the equation. It is also important to know when a needs assessment is warranted.

Often vendors complain that conducting a needs assessment takes too much time and lengthens the sales cycle. What isn't taken into consideration is when a needs assessment is valuable and an added tool to the sales process. Following is a list of situations in which a needs assessment becomes a valuable tool for the sales process:

(1) when a client is interested in teleconferencing, but does not understand the technology and is not ready to make any decision,

(2) when a client has decided on a technology, but has not decided on how the technology will be best used,

(3) when a client has already installed teleconferencing equipment, but finds it is not being utilized,

(4) when a client wishes to expand a teleconferencing system, but has not kept any records on the usage of the current system and now must justify expansion to upper management,

(5) when a client has one of your existing systems, but doesn't seem to use the system and isn't thinking of growth.

After The Needs Assessment

Once a client understands the corporate communication needs, it is important for the vendor to help the client meet those needs. The features and benefits of a vendor's product should be wrapped around the identified needs of the client. For instance, if the needs assessment indicates that the client needs to see others at distant locations to share management ideas, the product should be positioned as being able to meet that specific need. No longer should the product be pushed without identifying how it meets the customer's needs. Another way to show how the product meets needs is to discuss applications of the product in other client situations. "Mr. Customer, ABC Company needed to share production problems with its plant sites. Our product was recommended and accepted to meet that need. I think it could also be used to communicate your production problems between your plant sites."

With some preparation and thinking about this new approach to sales, increased sales should result. Perception is the key to success and proper prior planning will help the client see you as the preferred vendor.

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RASTI

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arguments will, of course, remain at least in those areas where "performance" is as much an artistic preference as an objective parameter. However, with respect to speech intelligibility much of this confusion can now cease. Individual preferences will still exist—we all have varying degrees of hearing impairment and speech recognition. But, for those who wish to clarify the situation, a practical tool is now available.

NOTES

1 "Methods for the Calculation of the Articulation Index," American National Standards Institute, 3.5 (1969).

2 "Report on the RASTI-Method for the Objective Rating of Speech Intelligibility in Auditoria," *IEC Publication 268: Sound System Equipment Part 16.*

3 H.J.M. Steeneken and T. Houtgast, "A Physical Method for Measuring Speech-Transmission Quality," *JASA*, 67 (1980), pp. 318-326.

4 John Anderson and Torben Jacobsen, "RASTI Measurements in St. Paul's Cathedral, London," Bruel and Kjaer Instruments Applications Note BO 0116-11.

5 Thomas R. Horrall and Torben Jacobsen, "RASTI Measurements: Demonstrations of Different Applications," Bruel and Kjaer Instruments Application Note BO 0123-11.

6 T. Houtgast and H.J.M. Steeneken, "The Modulation Transfer Function in Room Acoustics," *Bruel and Kjaer Instruments Technical Review*, 3 (1985).

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THEORY & APPLICATION

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which was not possible in the past under field conditions.

One strength of the Sound Intensity Method is its ability to reject contaminating noise from other sources. Consider two loudspeakers which are situated side by side as shown in **Figure 3**. The Sound Intensity Method can measure the power output of the first speaker in the presence of the second one. If this is attempted with a sound level meter, the acoustical power output is overestimated because the measured sound pressure level at each point surrounding the noise source is the result of the combined sound pressure level of both speakers. The Sound Intensity Method measures the net flow in the direction of the microphones and rejects the effects of the second source if power is computed for the entire surface area surrounding the speaker.

To understand this complex idea, consider a simple situation as shown in **Figure 4**. Two sound sources produce plane waves which propagate in a direction parallel to the x axis. When source A is turned off, the sound power obtained around the entire surface ($A_1 = A_2$) is:

Equation 10

$$W_{ab} = -I_b \times A_1 + I_b \times A_2 = 0$$

where:

I_b = intensity vector at A due to B,
W_{ab} = sound power at A due to B.

It is important to recognize the physics and sign conventions for this case. The sign of the first term is negative because it represents a flow of power *into* the volume. The sign of the second term is positive because it represents the power flowing *out* of the volume. Because the power flowing into the volume is equal to the power flowing out, the total power is zero.

The total power flow due to source A is expressed as:

Equation 11

$$W_a = I_a \times A_1 + I_a \times A_2 = 2 \times I_a \times A_1.$$

When the total power is measured for sources A and B, the expression is: ($A_1 = A_2$)

Equation 12

$$W = (-I_b + I_a) \times A_1 + (I_b + I_a) \times A_2 = 2 \times I_a \times A_1$$

Even in the case where both sources are on at the same time, the sum of all intensity vectors and maintenance of correct sign conventions results in the correct result, which is the sound power of source A.

This same reasoning occurs with much more complex situations. The net power flow into the volume through one surface area is negated by the power flow out of the volume through a second surface area. The total sound power of one source can be measured in the presence of another source(s).

To observe other benefits associated with the Sound Intensity Method for measuring sound power, consider the case of a directional speaker in a semi-reverberant field, e.g. a gymnasium. If the output of the speaker is measured with a sound level meter, the sound pressure level will vary significantly with distance and direction. Attempts to quantify the power output are difficult and misleading. For example, if Q (0) = 10, and r = one-quarter meter, and a = .2, then L_p = 131 dB when L_p is measured along the axis of the speaker. Off axis, Q (90) = .1, and L_p is 112 dB. Obviously a major difference. What is the typical position? There is none! Similar calculations at four meters give an L_p of 108 and 103 respectively which are affected by the reverberant field. Attempts to quantify power output are very misleading if only a few select points are used around the source. If measurements are made close to the source, the directional effects may lead to the wrong sound power estimation. If measurements are made at large distances, the effects of the reverberant field will cause the sound power to be overestimated if the sound pressure level readings are used. What is the solution? Measure L_w with the Sound Intensity Method! Using proper scanning techniques over the *entire* surface area surrounding the source, e.g. a hemisphere surrounding the speaker, the total sound power is correctly estimated after intensity is weighted by area. The correct power output of the speaker can be measured in the field.

Summary

In acoustics, it is important to distinguish among pressure, intensity, and power. Power is the key quantity to be determined from which pressure can be computed. With the advent of the Sound Intensity Method, power can be measured accurately and quickly with a high degree of confidence. A powerful new tool has been placed in the tool kit of the acoustician.

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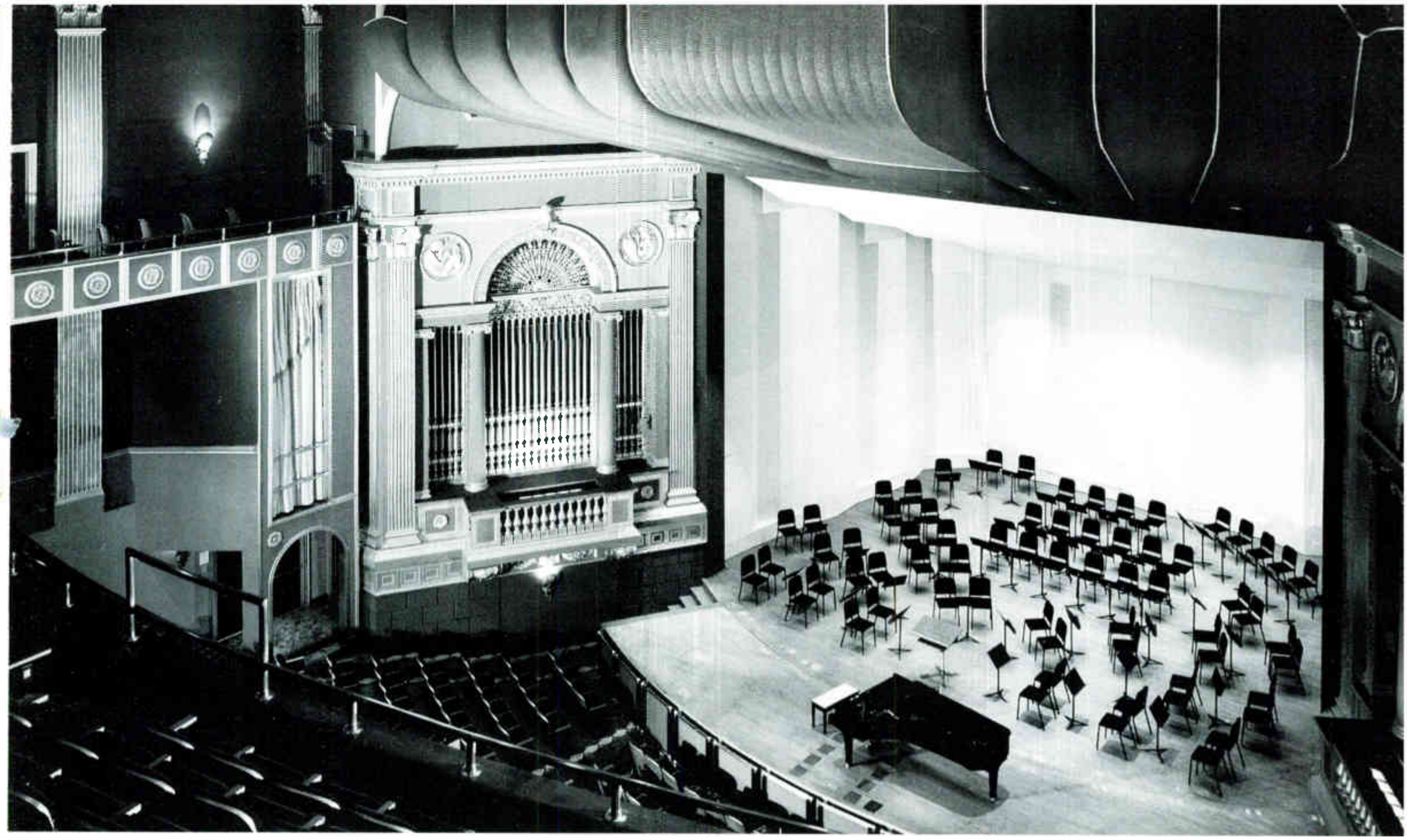

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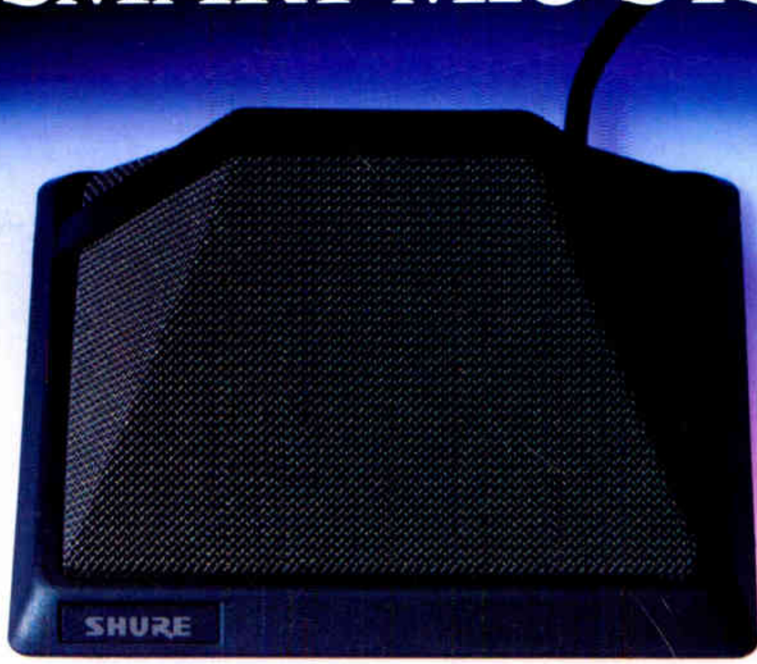

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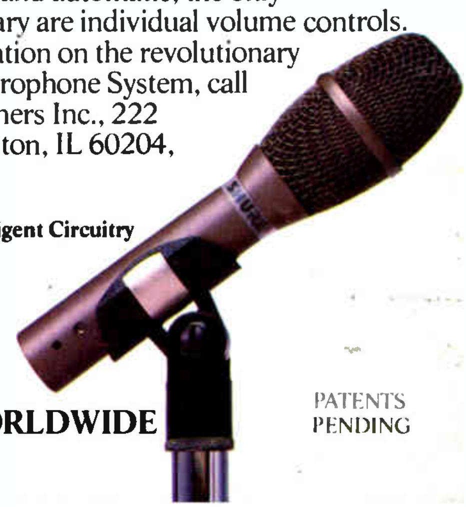

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