Lab Test: Oxmoor's DCA-2 and RC-16

SOUND & COMMUNICATIONS

FOR CONTRACTORS, SYSTEM MANAGERS AND SPECIFIERS NOVEMBER 1986

The Sound System in the Southern Baptist Church

The Origins of Mixing

Interpreting Time-Frequency Measurement

Intelligibility Theory: Applying the Test Methods

SOUND & COMMUNICATIONS

Volume 32 #11
November 1986

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DEPARTMENTS

ON THE COVER

The challenges of installing a sound system in one of the nation's largest churches is explored as well as how Stokowski helped bring mixing into the modern era. See pages 24 and 28 respectively.

FEATURES

COLUMNS

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Home Entertainment Magazine

by Jesse Kapholz

A Meeting of the Minds

Once again it is time for the industry's manufacturers, consultants, members of academia, engineers, dealers, and contractors to make their pilgrimage to Los Angeles for the Audio Engineering Society's Convention.

Through the years, the AES has been criticized, these criticisms are often uninformed attacks on the very essence of the industry and the reasons for the existence of the AES. And despite what we or many exhibiting manufacturers would like in terms of business or selling policies, the Society's gathering does not exist for that reason.

The AES was formed in 1948 for the purpose of gathering professionals engaged in the audio engineering field and its allied arts, and to disseminate technical information among its members. This year marks the 81st such gathering. A whole year has lapsed since the last convention, and as indicated by the program available at press time, there will be a multitude of exchange of technical information—not to mention new products and techniques.

But, what is the significance of all this hi-tech jazz to the sound and communications contractor? This is an opportunity where manufacturers, consultants, members of academia, engineers, dealers, and contractors will be on the same floor, at the same time, all there with the sole purpose of communication. This is a unique forum where the members of every discipline and practice can develop and maintain a first-hand appreciation for others' concerns and needs. Let's face the facts: T.I. or National are not about to develop a new chip, for example, specifically for the sound and communications marketplace. Therefore, if for simply this reason alone, it is valuable to participate in the endeavors of the Audio Engineering Society.

We are all constantly growing and expanding our markets through diversifying our basic talents to parallel technologies. Yesterday we were running 600-ohm loudspeaker lines; today we are running fiber optic and other types of data lines; tomorrow.... To prepare for tomorrow's marketplace, the academic and engineering sectors will report on new technologies about to be incorporated in new products and techniques.

Hone your skills and sharpen your pencils, at the AES you will hear from a *genuinely* representative cross section of the audio engineering industry's practitioners.

Papers to be presented will include the following subjects: perception, audio recording tranducers and sound reinforcement; audio measurements and instrumentation; architectural acoustics and listening conditions; audio recording; and audio recording and signal processing. Workshops, which are less formal, include among its categories: Future Directions in Professional Audio-A Forecast; Education in Audio—Does Testing Work?; Loudspeaker Cluster Design: The Art and Science of Equalization; Can we Talk? Production Intercom in the Entertainment Industry; Wireless Microphones—Why do they Work? Measurement and Instrumentation; Computers in Audio; Live Concert Sound; Microphones—out of the Studio and into the Real World; Loudspeaker Measurements and Tranformers in Audio.

These are topics in which *Sound & Communications* has a vested interest. We will be at the 81st AES Convention and will, in a future issue, report on the event. When all is said and done, we are happy to be at the 81st Audio Engineering Society Convention. See you in L.A.

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MANAGEMENT TEAM ACQUIRES BIAMP

Biamp Systems, which for the past 15 months was a wholly-owned subsidiary of Leupold & Stevens, a company best-known for its shooting sports products, has been acquired by the management team in a leveraged buyout. The new owners of Biamp are: Ralph Lockhart, president; Bob Doty, vice president of engineering; Ralph Tennant, vice president of manufacturing; Jerry Payette, vice president of finance; and two outside investors.

"Leupold & Stevens has concluded it needs to focus its management and financial resources on the core business since Biamp represented only a small portion of Leupold & Stevens', total revenues," said Payette. "The Biamp we have acquired from Leupold & Stevens is measurably stronger and better positioned than the one Leupold & Stevens purchased over a year ago," said Lockhart, "Leupold & Stevens' substantial investment has given Biamp the best equipped, most efficient manufacturing facility of its size in the industry; advanced computer aided engineering and automated test equipment tools, an aggressive advertising and point-of-sale brochure program, and a number of very successful new products including the MixPak Plus+ Series, the DJ3001 and the DJ5001 Disco Mixers, and two new product lines (the RackMax, a 16-channel stereo rack mount mixer and the XA100, a new family of stereo power amplifiers) which will be introduced at the Audio Engineers Society Show."

"We intend to continue and strengthen...Biamp's traditional role of furnishing innovative, reliable, high performance audio equipment to the professional audio, sound contractor and music instrument markets. We are committed to the business and committed to being an outstanding supplier," Lockhart added.

ELECTRO-VOICE & ALTEC COMBINE MARKETING EFFORTS IN CINEMA SOUND

Bob Pabst, president of Electro-Voice, Inc. has announced that the newly designated Mark IV Cinema Systems will include Altec Lansing Voice of the Theatre electronic and acoustic products and Electro-Voice Theater Sound Systems acoustic products. According to Janine M. Fromm, sales manager for Mark IV Cinema Systems, the existing account base for both product lines will be serviced by the Electro-Voice network of independent manufacturers' representatives. "Theater dealers will find Mark IV Cinema Systems an exciting marriage of product lines," Fromm stated. She also said that the next best thing about the joint venture was that "a broad array of acoustic and electronic products will now be available from a single source, insuring prompt delivery, easy communication, and extensive physical distribution and improving our ability to better serve the theater industry."

FORMER MOTOROLA PRODUCT LINE ACQUIRED BY HITK

High Technology Capital Corp., the nation's largest publicly-owned business development company, has acquired VCS, Inc. a manufacturer of custom integrated security systems and closed-circuit television (CCTV) equipment. VCS, formerly a Motorola product line, became an independent concern in 1981. Today, the company serves as a supplier of CCTV equipment to Motorola. Established as a subsidiary of Motorola, Inc. in the early 1960's VCS was the sole manufacturer of Motorola CCTV equipment. VCS is currently headquartered in a 32,000-square-foot manufacturing facility in Carol Stream, IL. The introduction of a new national network of independent service shops and dealership locations throughout the United States will help to further broaden VCS' customer base, according to the company. Matthew J. Tummillo, formerly operations manager at Motorola from 1978 to 1981 and now vice president and general manager of VCS, will serve as chief operating officer of the newly-acquired company.

BEIJING RECORDING '86 FIRST AUDIO SHOW IN CHINA

Beijing Recording '86 represented the first organized audio trade show in mainland China. The International Technical Interchange and Exhibition on Professional Sound Recording Equipment was organized by the Beijing Acoustic Society and took place at the Minzu Exhibition Center from August 30 to September 5 in Beijing, the capital of China. Many international manufacturers were represented by six distributers: Studer Revox, Wo Kee Engineering Ltd., Audio Consultants Co., Auvix-Asona GMBH, Advanced Communication Equipment (ACE), and the Power Source Development Ltd. A demonstration of the Genelec monitor speakers and RPG Diffusors was organized by Bingo Tso at the China Central Broadcasting Center. Present were academic acousticians. This was followed by a tour of China Records recording studios.

CREATIVE MARKETING GROUP APPOINTED ARIES IMPORTER

Creative Marketing Inc. of Los Angeles has announced its appointment as exclusive import agents for the "Aries" line of professional audio mixing consoles, manufactured in London, England. The line presently consists of two models, a 16/8/16 and a 24/8/16. The Aries consoles are priced at $5,400 in the 16 input configuration and $6,995 for the 24 input. Three additional models will be available this fall.

ICIA SAYS NO TO PAYMENT IN 80 DAYS—FEARS FOR SMALL COMPANIES

The International Communications Industries Association is campaigning to stop a new plan which will allow U.S. agencies to pay bills in 80 days. The 80 day proposal is part of a plan issued by the Department of Defense, National Aeronautics and Space Administration, and General Services Administration in which they want agencies to have five days to receive goods, 30 days to process receiving papers, 30 days to process payment papers, and 15 days to issue the check. "This adds up to 80 days, more than twice the number of days allowed by the current law," commented ICIA Legislative Committee Chairman John Moore, Jr. Contending that a 1982 Act of Congress set the payment standard at 30 days, ICIA argues that the 80 day standard would drive small communications businesses away from the Federal market and would label the U.S. as a bad business partner. The national campaign to defeat the 80 day proposal is being led by ICIA staffer Kenton Pattie who leads the 28 associations that belong to the Coalition for Prompt Pay.

BENJAMIN CONSOLIDATES LINES IN NEW FACILITY IN PLAINVIEW, NY

Benjamin International, Inc. has consolidated its three lines in a new facility at 1460 Old Country Road, Plainview, NY 11803. For the past three years Benjamin has been importing and distributing the Model Acc-15 Automatic Cassette Changer. The sales and service for this model as well as the other two Benjamin lines—the Web Detection line of Vehicle alarms which is imported and distributed by Benjamin and the complete line of Bulk Tape Erasers which is manufactured by Benjamin—will be handled at the new location.

UNEX MOVES TO WESTFORD, MA

Unex, A Dynatech Company has announced that it has moved to a new address. The company is now at 3 Lyberty Way, Westford, MA 01886. The telephone number remains the same.

Warranties

YOU'RE AS GOOD AS YOUR WORK

When it comes to warranties most of us immediately think of the "five years or 50,000 mile" guarantees car dealers boast about in commercials. But, when it comes to sound systems, what kind of mileage can your client expect from the installation you do for him?

Most sound and communication contractors today warranty their work and the product installed for at least one year. If you're doing a job which was awarded through the bidding process, you can practically guarantee that the consultant who designed the system is going to require that you warranty the installation.

Although most warranties, which are included as part of the contract, bind the contractor to repair or replace any defects in a system during the warranty period at no cost to the client—warranties can also be turned into profit for the contractor. A warranty is a primary feature when selling a system. Since a warranty more or less tells the client that you, the contractor, will *guarantee* every aspect of the system from the product installed to the installation itself (provided it does not show abuse), the warranty implies to the client that you are confident that the system will perform correctly. Once a warranty has expired, a client still will want to enjoy the service and attention he got from the contractor when his system was *covered.* And you can offer him that service for a fixed price under an *extended* warranty or service contract.

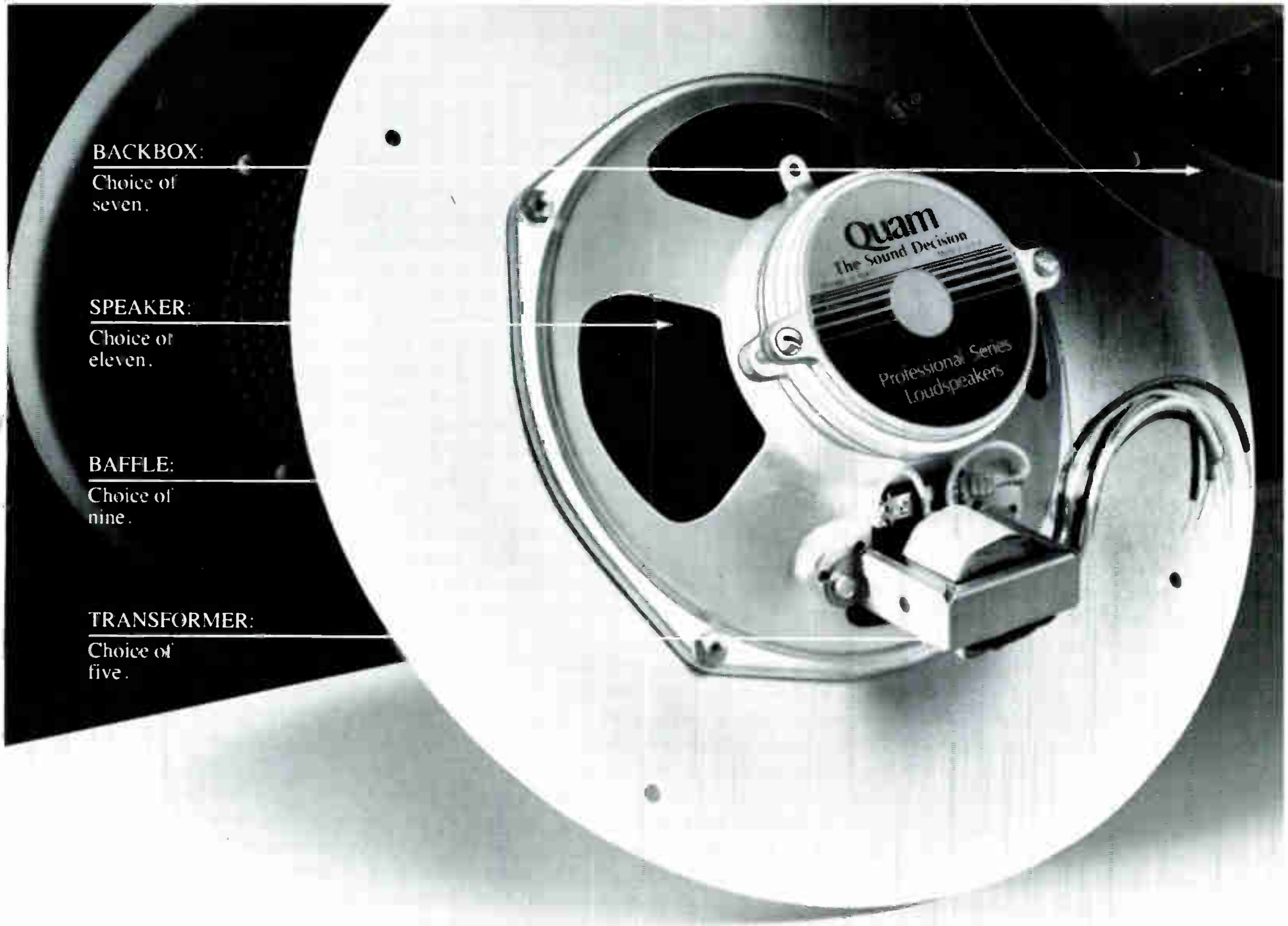
Extended warranties guarantee steady income for the contractor, as well as function as a medium for the contractor-customer relationship. The extended warranty, in a sense, locks the customer into dealing with you. This opens the door for you to sell the add-ons, modifications, and updates. It also makes you the number one source if new work comes up.

Estimating Rates

The end user's cost for an extended warranty varies with each system. Obviously, an extended warranty for a system which is large and fairly complex—like a hotel conferencing system with background music, PA, and teleconferencing—would be higher than that for a simple background music system. When establishing a rate for an extended warranty on any system, the following variables should be considered.

- The size and magnitude of the system.
- The complexity of the system. The more complex the system, the longer it will probably take to diagnose the problem.
- The life-expectancy of the equipment installed, which is based on the following.
  —The various manufacturers' warranties on each of the units.
  —The amount of mechanical parts involved in the system. Mechanisms, such as cartridge and tape machines, tend to break easily and wear faster.
  —The frequency of system use. How often is the system being operated? The more a system is operated, the more likely it is to encounter problems.
  —The number of and experience of system operators.
- Location of the system. Is it an index or outdoor system?
- Travel time. A system within your own city or town is easier and less expensive to get to than that which is across the state or across the country.
- Length of time of extended warranty.
- Abuse and misuse of a system should *not* be covered under warranty.

Extended warranties not only provide contractors with extra income, but they tell customers that you're willing to stand behind your work. If you don't already offer warranties, you should seriously consider doing so.



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Part II: The Design Phase

by Marc L. Beningson
Jaffe Acoustics, Inc.

COMPLETING FINAL DRAWINGS

In last month's column, the architect and his team of consultants and engineers were awarded a major project and generated schematic design documents presenting the design intent based on their specialties.

Even when a design team has worked together previously on similar projects, there will be some disagreement among the members of the team when the design development begins. The architect wants an impressive and attractive building; the theater consultant emphasizes good sight lines and ideal lighting positions; the acoustical consultant requires the proper distribution of absorptive, diffusive, and reflective surfaces to satisfy the acoustic criteria for the intended program;

***There is a major difference between a* design *and a* specification.**

and the owner wants the budget to stay intact.

On occasion, conflicts develop. Because the requirements for sound systems and acoustic criteria are least known, the sound system designer and the acoustical consultant often take what seems like more than their share of abuse. The architect is disappointed that there will be all those *ugly* speakers all over the proscenium and stage, and he has trouble with all the extra concrete and sheetrock layers required for sound isolation. The theater consultant understands the acoustic and sound system requirements, but is fighting his own battles for lighting positions, pit lifts, rigging, and dimmer rooms. The owner wants to know why a "house mix position" is going to take away a dozen of his highest priced seats. More importantly, the owner may not have included in initial budget estimates the additional costs of sound isolation walls and doors, duct silencers, or a sound system that fully meets the requirements of the intended program.

A good consultant knows when compromise is needed and when to stick to his guns; and a good architect learns to trust his consultants' expertise and experience, and finds a way to balance all the requirements—hopefully within the budgetary limits. A good look at the program and input from the owner and user groups will usually cast sufficient light to generate the priorities for the project. Design development ends when a cohesive plan for the project has evolved into its final form, and the design team is, for the most part, in agreement on how the plan will be executed.

Each consultant begins to generate working drawings in the final portion of the design phase. The object is to generate specification documents for each of the various subdivisions of work necessary to construct the project as designed.

In a way, the physical acoustic portion of the work is simpler for the consultant. His sketches, recommendations, and designs are integrated into working drawings by the architect. Thus, the physical acoustic work is specified as part of the workscopes of the various trades—drywall, concrete, fabric wall, window, and acoustical tile ceiling installers.

From the sound system design however, a complete specification must be generated from the design during the working drawing portion of the design phase. There is a major difference between a *design* and a *specification.* A design is simply a block diagram and an equipment list. A specification is more like a recipe, it lists not only the equipment, but explains how to put all the parts together. Part I includes a system description and sections explaining the complete workscope (particularly what is not included), procedures for and requirements for shop drawings and other submittals, how substitutes or alternates may be dealt with, job conditions, quality assurance, and guarantees. Part II includes the equipment list. Part III explains how the installation is to be executed, conduit and electric power requirements, methods and practice of wiring and grounding, initial adjustment, documentation requirements, and acceptance testing procedures. Location drawings and detail sketches accompany the specifications to form a complete package for bidding.

Coordination among the members of the design team is essential, so that the electrical specifications clearly spell out the responsibilities of the electrical contractor regarding sound system requirements. It is very easy for enclosures, backboxes, wire installation, and other items to be assumed by both contractors to be in the workscope of the other. Similarly, the architect's drawings and specifications must demonstrate to the general contractor and other subcontractors any requirements to accommodate sound system devices. This involves everything from acoustically transparent surfaces to control room countertops and portable house mix positions, ceiling speaker locations, and the like. The sound system designer and mechanical contractor must also review each others drawings to ensure that ducts, grilles, and sprinkler heads don't interfere with speaker locations.

Generally, after each member of the design team has reviewed the complete set of working drawings for the entire building, a coordination meeting is held to deal with any final problems. After the final set of corrections are made to the specifications and drawings, the entire package is put together with the owner's general conditions package to form a complete package of construction documents. The design process is over, and the construction phase of the project begins. At this stage contractors enter the picture and the specifications must be interpreted by companies whose profit rests on their ability to determine what the specifications really mean. And the fun begins.

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by Daniel Queen
Daniel Queen Associates

POWER SYSTEMS

As consumers, we take for granted the power supplied from a wall outlet. Few of us would suspect this usually reliable convenience to be a potential source of noise and instability problems in a sound system. Unfortunately, by the time a completed power system is identified as a source of problems, correction costs can be high. The proper design of the main power supply of a sound system prior to installation adds little cost but much satisfaction.

Instabilities from power lines usually originate from two sources. First, variations placed on the power line by external phenomena outside of the site. And second, variations caused by equipment on site when connected to the power line.

Such variations become problems due to conflicting demands on sections on the power system and confusion of responsibilities within the system. To avoid such unpleasantness, we take a lead from the Founding Fathers and invoke the separation of powers.

Different circuits must be made responsible for lighting, for mechanical equipment, and for the sound system. It is advisable also to separate circuits used for computer equipment and video. By bringing these different types of circuits independently from the service entrance panel of a building, one minimizes the variations which each can impose on the other.

Power line variations fall into three major categories: long term variations in the supply voltage; short term variations such as surges, dropouts, and spikes; and transient noise which may be intermittent or steady state.

The American National Standards Institute publishes ANSI C84.1-1982, which details the high and low voltages that may be expected from a *standard* power service. If according to the specifications for the equipment in use, the appropriate standard voltage limits may be tolerated, no conditioning need be applied to correct the voltage on the power line. Most audio equipment utilizing regulated power supplies is designed to accommodate the worst conditions on a standard power line. However, even when specified to operate over the full range of line voltage, some equipment will not, for example meet its maximum power output during a low line condition.

Similarly, when connecting recording and video equipment, care should be taken to examine specifications very carefully to determine the amount of regulation needed on the line.

Special care must be taken to make the type of regulation chosen compatible with the equipment being used. Four general types of regulators are available: transformer tap changers, ferroresonant transformers, SCR controllers, and continuously adjustible autoformers.

For sound and video equipment, only the continuously adjustable servo-controlled autoformer is advisable. Other types can introduce transient noise and variations which may be intolerable to the equipment. Regulators utilizing tap changers and ferroresonant transformers are usually suitable for use with computer equipment. Regulators using SCR controllers should ordinarily be used only for lighting and heating circuits.

Autoformer regulators are designed to handle only long-term voltage variations. Transient variations, such as surges and momentary dropouts, must be handled by energy storage or dissipation devices such as surge protectors. Such devices may be purchased alone or as part of power conditioning equipment.

The simplest form of surge protector is the varistor, which is able to clip spikes that occur above three times the line voltage. Protection at lower voltages is provided by various forms of breakdown diodes. Gas discharge tubes are often used for high voltage transients. Each type has advantages and disadvantages which are utilized appropriately by power conditioner manufacturers. Attention must be paid in specifications not only to the voltage and current levels of protection, but also to the number of events the conditioner

(continued on page 49)

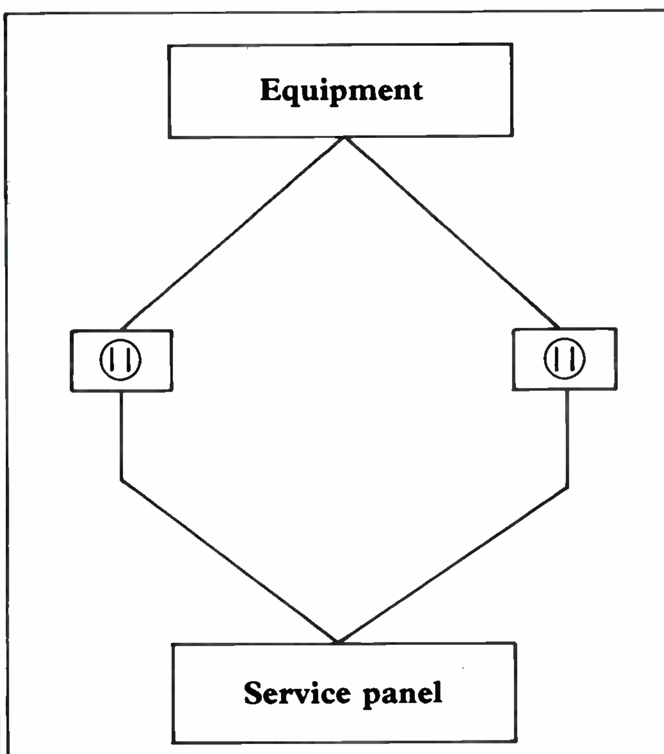

Figure 1. Common grounds can act as inductive pickup loops when improperly installed.

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Intelligibility Theory

Applying the Test Methods

by Steven J. Orfield
Orfield Associates

While there is an overwhelming desire within both the sciences and the social sciences to develop comprehensive theories within which to couch explanations of complex phenomena, philosophical analysis has long ago taught us that there is often more to be lost in the gross generalizations produced by theoretical models than to be gained by the narrow value of the explanations.

> *". . . the appropriateness of an explanation is determined not by the phenomenon it seeks to account for, but by the question it seeks to answer." 1*

Intelligibility testing theory tends to exhibit that desire for generalization at the cost of explanation. As intelligibility testing is dealt with in this discussion, a *narrow* definition is assumed:

> *"Intelligibility is the accuracy which the transmission path allows for the transmission of the acoustic signal in the time and frequency domains." 2*

With this definition in mind, a discussion of different methods of evaluating the "impulse response" of the communication path will follow, with an emphasis on current test technology. A discussion of the broader issues will follow an explanation of test technology.

Four methods of evaluating the relative change in signal distortion of a communications path that were discussed previously are:

1) Word Score Tests
   (ie. ANSI S3.2 1960)
2) AI Tests
   (ANSI S 3.5 1969)
3) Alcons Tests
   (Variations of Peutz/Klein equations)
4) RASTI Tests
   (Draft IEC 268, Part 16)

(I have not included early-to-late ratios, although they are also under much study for intelligibility.)

I will first review the standard test format for each of these and then review some interpretations of each test format.

PB Word Score Tests

The explanation that follows is based on the USA Standard for the Measurement of Monosyllabic Word Intelligibility (ANSI 3.2 1960). This test is based on a set of single syllable words that have been selected due to their representation of different speech sounds based on the frequency of occurrence of those sounds in the language. There are 20 different lists of words that are inserted in the same *carrier sentence,* "Would you write ____________ now."

The carrier sentence should be spoken as a *simple declarative sentence with no unnatural stress on any word.* The listener writes down the word that he thinks is heard, and the resultant word is judged for correctness by its correct indication of the sounds in the word uttered, without regard to appropriate spelling. The final test score is based on the percentage of words correctly noted.

Subjects for this test (talkers and listeners) are screened for hearing loss based on the criteria that they have 10 dB or less loss which is average *and* 15 dB or less loss at 250, 500, 1,000, 2,000, 4,000 Hz.

An approved audiometer test is required, and the talker must have no speech defects. A specific training process prior to formal testing is required by the procedure. The experimenter is free to select talker and listener required levels based on a recommended word rate of 15 key words per minute. Recordings of these word lists can be used in lieu of a live talker, as long as the limits noted in the test format take place. Since there are many uncontrolled variables in this type of experiment, it is explicitly a relative test of the effectiveness of communications paths.

In addition to this specific test format, there are many other talker-listener tests that have been used in the evaluation of relative speech intelligibility, as noted by the ANSI Articulation Index standard:

- 32 PB words
- sentences known to listeners
- sentences unknown to listeners
- 256 PB words
- rhyme tests
- 1,000 PB words
- 1,000 nonsense syllables

Each of these listed test formats is considered to exhibit a different level of listener difficulty, and very specific Articulation Index scores have been attributed to different levels of relative success on each test. This test can be performed using any transmission system or can be used live in a non-amplified test of a space.

Since all intelligibility testing is verified by some form of talker-listener test, these tests are clearly the most important and fundamental tools in the evaluation of relative communication path intelligibility.

Articulation Index Tests

This discussion of the Articulation Index procedure is based on a standard format entitled, "American National Standard Methods for the Calculation of the Articulation Index," (ANSI).

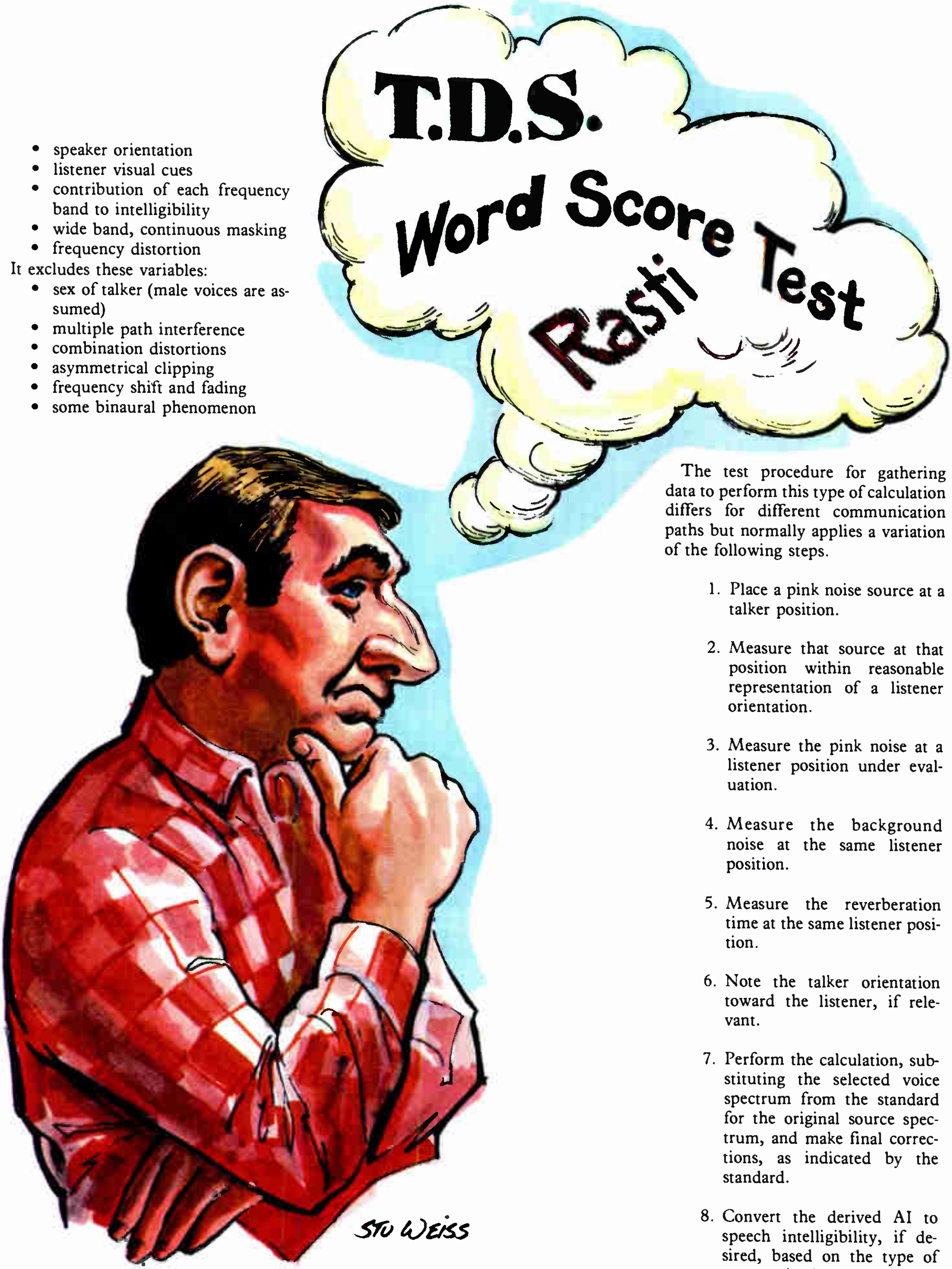
The Articulation Index procedure is not strictly a test procedure, but is rather a procedure for calculational consideration of a set of acoustical variables, including:

- average and peak voice level
- background noise level
- reverberation time
- transmission loss of signal in a number of frequencies (normally 15 one-third octaves)

- speaker orientation
- listener visual cues
- contribution of each frequency band to intelligibility
- wide band, continuous masking
- frequency distortion

It excludes these variables:

- sex of talker (male voices are assumed)
- multiple path interference
- combination distortions
- asymmetrical clipping
- frequency shift and fading
- some binaural phenomenon



The test procedure for gathering data to perform this type of calculation differs for different communication paths but normally applies a variation of the following steps.

1. Place a pink noise source at a talker position.
2. Measure that source at that position within reasonable representation of a listener orientation.
3. Measure the pink noise at a listener position under evaluation.
4. Measure the background noise at the same listener position.
5. Measure the reverberation time at the same listener position.
6. Note the talker orientation toward the listener, if relevant.
7. Perform the calculation, substituting the selected voice spectrum from the standard for the original source spectrum, and make final corrections, as indicated by the standard.
8. Convert the derived AI to speech intelligibility, if desired, based on the type of communication.

This same standard can be performed based on theoretical calculations, as has been done in many cases, including that of the open plan offices. It is also popular within the field of acoustic consulting.

It is important to note that AI does not convert directly to intelligibility, as very high intelligibility is derived from a modest percentage of speech information. The comparison scale usually sited for conversion of the corrected AI calculation based on conversational speech is:

Articulation Index	Speech Intelligibility
1.00	100%
.80	99%
.60	98%
.40	93%
.20	50%
.10	18%
.05	8%

Articulation Loss of Consonants

By far the most popular procedure for the evaluation of speech intelligibility within the audio design community is that of calculating the Articulation Loss of Consonants, a procedure made popular in the 1970s by Peutz and Klein. This procedure begins to address both previously considered variables and a more complex set of measurement parameters taken from the field of general architectural acoustics.

- signal-to-noise ratio
- source directivity
- room volume
- number of sources
- direct versus reverberant field
- reverberation time
- listener distance
- source coverage

While the Articulation Index considered the communication path under many criteria, it was not nearly as source, listener and room specific as the ALcons calculation.

The information basis of the ALcons calculation can be either actual testing or theoretical prediction and can be gathered via many test methods, including real time analysis, narrow band FFT analysis, and time delay spectrometry. There is much discussion as to what methods should be used to derive specific data for input into this calculational base, such as:

- What is the definition of signal content in the signal-to-noise ratio (direct vs. reverberant)?

- What reverberation time char-

In Review

Syn-Aud-Con's Conference on Intelligibility

In the audio field, there is very little beginning level education available and also a limited amount of continuing education, made up principally of seminars set up and paid for by the manufacturing community. One additional offering is the Syn-Aud-Con organization of California. This small group tends to support some specific theories of audio design and testing technology, and it offers seminars that are partially fee based, partially supported by sponsoring manufacturers, and partially supported by user memberships.

With a strong following in parts of the audio community, there is enough combined worship and mythology surrounding the Syn-Aud-Con organization, that I decided one year ago to join Don and Carolyn Davis' group of audio practitioners, not really knowing what I had or had not accomplished. I know of quite a few people on both extremes of the continuum with regard to this group, and the benefits of membership seemed to range from curiosity to education.

Having recently received a notice of a four-day intelligibility seminar in Chicago, described via some pretty enthusiastic language, I registered and found myself in Chicago for two of the four days, listening to the likes of Don and Carolyn, Dr. Peutz, Richard Heyser and the staffs of Techron and Bruel & Kjaer.

The premise of this seminar was that of examining three large room sites in the Chicago area via four test methods to demonstrate the use of these test technologies and their benefits and the intelligibility differences in reverberant environments between High-Q, Medium-Q, and Low Q devices.

The test technologies were Alcons, RASTI, dual channel FFT equalization, and word score tests.

The actual testing and collation process was quite cumbersome, and I suspect that many non-practitioners of these tests did not have a reasonable sense of the logic of the technologies or their use. While the methods of this seminar were less than scientific, the results were as predicted, thus demonstrating the underlying view of the group that Q is a valuable prediction variable in sound system design. A discussion followed at some length, including a complex theoretical dissertation from Richard Heyser, not bearing on intelligibility.

Had this seminar provided no more than this, I would have been somewhat disappointed, as I felt that many of these individuals had far more to offer than was apparent at this meeting. Fortunately, my most memorable experiences were all in private discussions with the speakers and attendees.

Having read much of the Syn-Aud-Con information related to the prohibition against the use of double columns in large rooms, I asked Dr. Peutz what he would do to resolve a poor quality double column system at one of the Chicago sites. To my amusement, he suggested the use of two additional columns, noting that a central cluster system would be unnecessary. (He explained that in Holland churches spent little on systems.) Additionally, I asked Dr. Peutz what was his basis for the equalization of sound systems, and he indicated that he equalized for the accuracy of sound and then corrected for intelligibility. He noted a philosophical distaste for equalization to enhance "quality." This provides a clearer understanding of his view of the priority of intelligibility as an issue.

A discussion with Don Keele of Techron introduced me to one of the brighter individuals that I have had the pleasure of talking with as he gave some insight into the directions that the TEF folks are headed. I have long had the view that the TEF was an "insiders' machine," and Don is taking some clear steps in attempting to separate its theory from its use.

John Barchan and Marty Alexander from Bruel & Kjaer expressed a very open view of the testing process, even though their RASTI meter has not been a favorite of Syn-Aud-Con, and they had much to say concerning directions in which acoustic testing was moving.

Helmuth Kolbe of Switzerland, one of the only users of time-delay spectrometry for architectural modeling, was very helpful in providing information concerning his modeling practice. With many additional stories of this type, the benefit of the Syn-Aud-Con organization begins to be apparent; for the experienced designer of sound systems who is equipped to perform sophisticated testing, there are a large number of experienced and clever people gathered in one place who are interested in discussing the field and sharing their experiences. Regardless of one's view of the specific philosophy of Syn-Aud-Con, this opportunity to share information with a group of generally non-defensive professionals is delightful and educational.

—Steve Orfield

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acterization is most advisable (RT/60, RT/30, RT/20, RT/10, RT/5)?

- What is the proper method of establishing Q?
- Which theoretical reverberation calculation is advisable based on distribution of room absorption?
- How do you deal with dissimilar sources?
- What frequency range should be under consideration?

Once the ALcons has been gathered or calculated, it is normally compared to some preconceived criteria, based on the need for intelligibility control in the space. The most common standard suggests that the ALcons level should be a maximum of 15 percent, with lower levels of loss for more critical definitions of needs.

Among practitioners of this standard, a clear measurement preference is being established in favor of the use of time delay spectrometry and, specifically, the use of the Techron TEF measurement system. A disk used for the calculation of ALcons on the TEF analyzer has been in circulation for some time, and a formal introduction of this disk will follow shortly.

Rapid Speech Transmission Index

The Rapid Speech Transmission Index (RASTI) standard is based on a test format, developed by Steeneken and Houtgast, entitled the Speech Transmission Index. This is specifically based on the application of the modulation transfer function, previously popular in the field of optics, to the analysis of signal distortion in the field of acoustic transmission.

This method called for the analysis of signal modulation at seven octave frequencies (125, 250, 500, 1,000, 2,000, 4,000, 8,000 Hz) under specific modulation frequencies. The reduction in modulation was then computed and averaged to determine an apparent signal-to-noise ratio, and a frequency weighted average of these S/N ratios resulted in the Speech Transmission Index. The RASTI calculation uses only two of these octave frequencies, 500 Hz and 2,000 Hz, to establish a close approximation of the overall STI values.

This test is normally performed via the use of the Bruel & Kjaer RASTI system, which includes a dedicated transmitter and receiver for signal generation and measurement. The transmitter is placed at the position of a talker (or is electronically inserted into the communication path) and the receiver is placed at the point of a listener. A direct reading and a set of variable readings results from this process. Recently, a very interesting alternative has been developed for the TEF system by Don Keele of Techron, which displays the data in curve format and also displays the resultant value. (This system is apparently going to be extended into a full STI format for alternate user selection.)

The RASTI method considers a number of variables, including:

- signal to noise ratio
- background noise level
- early decay time
- frequency distortion
- time distortion

The measurement or calculational process has certain limits.

- It assumes a linear transmission.
- Pure tones are not considered.
- It assumes constant noise floor.
- Reverberation time should not be strongly dependent on frequency.

A system for converting the RASTI values to speech intelligibility values has been suggested, as noted below, along with a conversion between RASTI values and ALcons values:

RASTI vs. Speech Intelligibility

0	.30	.45	.60	.75	1.00
Bad	Poor	Fair	Good	Excellent	

RASTI vs. Articulation Loss of Consonants

RASTI	ALcons
1.00	0.0
.90	1.3
.80	2.2
.70	3.8
.60	6.6
.50	11.4
.40	19.5
.30	33.6
.20	57.7

(Courtesy Syn-Aud-Con and Farrel M. Becker of Audio Artistry.)

Intelligibility vs. Accuracy versus Quality

A recent intelligibility seminar sponsored by Synergetic Audio Concepts brought together many practitioners in the field of audio and acoustic design and theory. A set of measurement sessions was performed based on the use of TEF ALcons analysis, Bruel & Kjaer RASTI analysis and Bruel & Kjaer Dual Channel Analyzer equalization.

This seminar was based on the same assumption that has been made in this article, that of the importance of the relationship between intelligibility and signal distortion, and it provided a demonstration of these concepts and discussions by many persons fundamentally involved with the concepts, including Dr. Peutz.

A somewhat broader view of the problem of intelligibility confronts the consultant or contractor when a sound system is installed and tuned. There is much information and experience that suggests three basic evaluative criteria which are important within the field of sound system design: intelligibility, accuracy, and quality.

Anyone who has ever adjusted a sound system knows that the highest level of adjustment for intelligibility at the lowest signal level provides neither optimal quality nor optimal accuracy. There are as many views of sound system adjustment and equalization as there are practitioners, and there is a clear need to characterize the relationship between these three variables for different types of sound systems, audio material, listening spaces, and listener preferences. While there has been much research into each of these areas, the need for coordination of this information is great.

Sound System Performance Definitions

Additionally, there is a substantial need to work toward performance definitions of sound systems that include listener and speaker variables, both related to the message being transmitted and to the perceptual variables inherent in the speaking and listening process. It is ironic that there is far more variation in actual intelligibility accounted for by variables not considered in any of these theories than in many variables currently considered. In using variations within the Articulation Index standard as a base, the following maximum variations in each of these variables are accounted for:

Variable	Maximum
1. speaker clarity	1.00 or >
2. message difficulty	1.00 or >
3. importance of message	1.00 or >

(continued on page 34)

INTERPRETING TIME-FREQUENCY MEASUREMENTS

by E. Curtis Eichelberger

Advances in measurement instrumentation and digital processing techniques have opened up new possibilities in audio measurements. It is now relatively easy to simultaneously measure in both frequency and time domains, which can provide great insight into the performance of audio components and systems. As is with any new technology, these measurements can also be confusing. The following provides an overview of time-frequency measurements and their interpretation.

Frequency Measurements

The classical approach to studying electro-acoustic devices has been to measure the frequency response. The concept of frequency is usually explained by a discussion of musical tones and of mechanical resonances—both of which are a function of frequency. Through these familiar physical phenomena most of us come to understand frequency analysis.

Through the work of Jean Baptiste Fourier, it has been revealed that a periodic signal, such as the sound pressure radiated by a vibrating string, can be completely described by the superposition of an infinite sum of sinusoids with the proper phase relationship. The result is the discrete or line spectra, where the magnitude and phase of each line is uniquely defined. This principle can be further extended to non-periodic signals such as random noise and transients by letting the period approach infinity, resulting in a continuous frequency spectrum.

The classical method of performing frequency domain measurements is with swept sine excitation. The recent introduction of low- cost Fast Fourier Transform (FFT) analyzers has also made feasible the use of other types of excitations such as random noise, tone bursts, and impulses—even the musical program material itself.

The power and speed of frequency analysis for solving some kinds of engineering problems has led to its widespread use. This is particularly true of telephone transduction devices and transmission line problems. Much of the pioneering work in the audio field was performed by telephone engineers—primarily at Bell Laboratories. These pioneers made great use of frequency analysis, and its use is deeply embedded into the audio field. But through the efforts of modern pioneers, such as Richard Heyser, J.M. Berman, and L.R. Fincham, the audio engineer is becoming aware that some acoustical problems are more easily detected and evaluated in the time-domain.

Time Measurements

Time measurement needs little explanation. If a short duration signal is input to an electro-acoustic device, we hope to see a faithful reproduction of that signal at the output. The faster the rise time and shorter the duration of the signal, the more difficulty the device under test has in reproducing the signal. The device being tested may be slow in achieving full output and the output may continue long after the excitation has ceased.

Just as the response of a system can be completely described by the frequency domain, it can also be completely described in the time domain by the unit impulse response. This is the response of the system to an input signal of unit energy over an infinitely small duration. What is interesting is how this impulse of energy is spread over time at the output.

Infinitely short duration impulses, of course, cannot be produced; but, good approximations or step functions can. As long as a certain bandwidth limitation of say B can be dealt with, then the impulse need only be less than $T = 1/B$ in duration. In fact duration should be limited to T so that all of the input energy can be concentrated in the frequency region of interest.

Unfortunately, even with gating, the response quite often is buried in noise, reverberation or resonance decay. All physical systems have dynamic range limitations and the input signal usually must be limited to the upper end of this range. This also precludes getting a lot of useful information about the distortion of the system as a function of amplitude.

Many of these signal-to-noise problems have been overcome by a recent measurement technique called Time-Delay-Spectrometry (TDS) [1]. TDS offers exceptional performance in situations where immunity to background noise is essential. By generating a linear frequency sweep (sweep rate S in Hz/sec.), TDS converts a time delay into a frequency shift. Time delays now become frequency offsets. A bandpass filter of bandwidth B (in Hz) in the frequency domain now becomes a window in the time domain. For a given sweep time, Ts, the frequency range, F, and the time range, T, the equations are as follows:

$T = B/S$ and $F = S*Ts$

The impulse response can also be obtained by the inverse transform of the frequency domain data. With the help of the Hilbert Transform, the time signal can be treated as a complex valued function (just like the frequency domain) and the magnitude and phase of the time signal can be computed. The magnitude of the time domain signal is called the Energy-Time-Curve (ETC). The ETC is a non-negative function and it can be viewed as the energy flow from the system under test. With editing in the time and frequency domain, the engineer can get very creative. One can, for example,

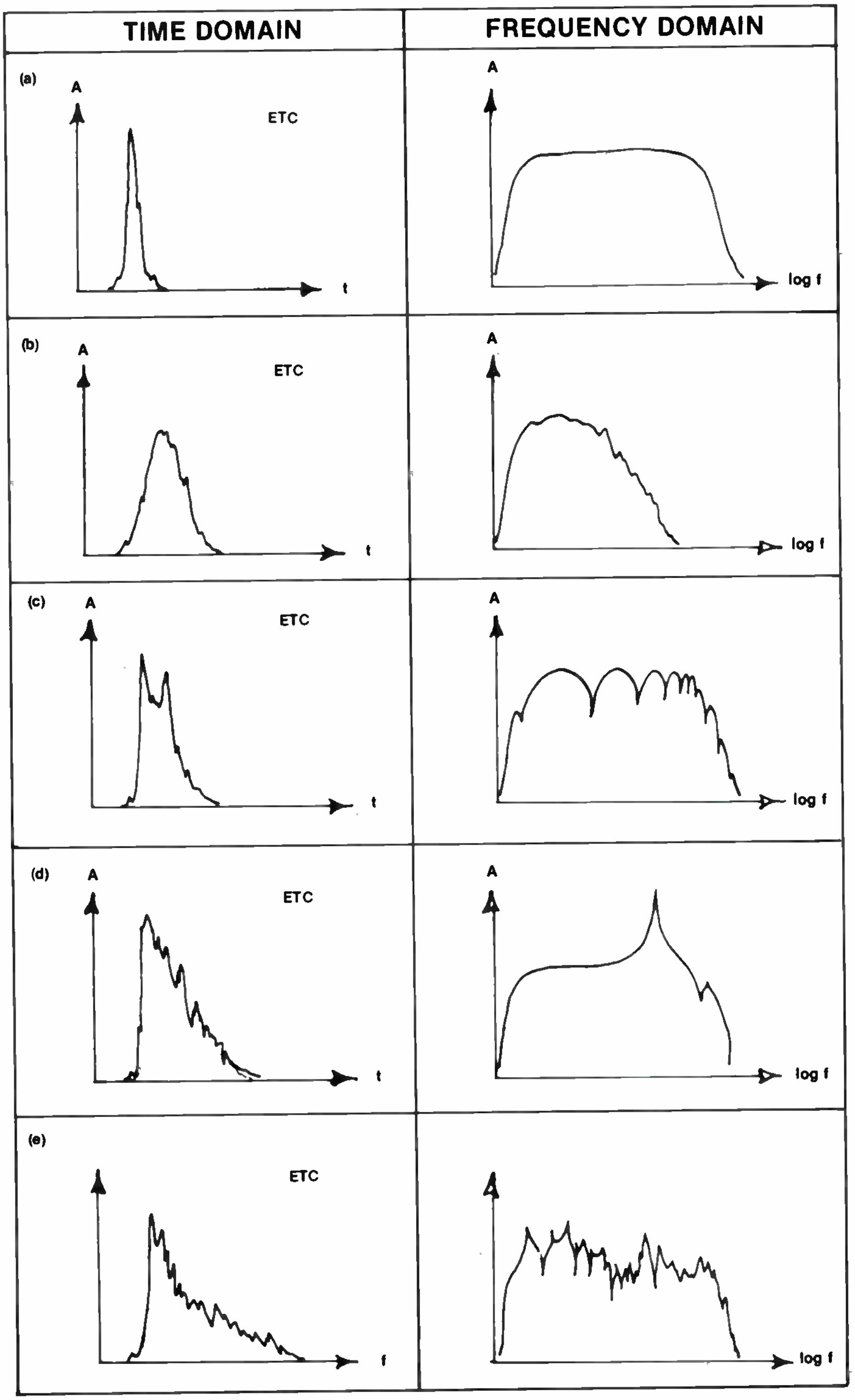

Figure 1: General characteristics of Energy-Time-Curves and the corresponding magnitude in the frequency domain. (a) Ideal system, (b) System with bandwidth limited by diffraction, (c) System with reflection, (d) System with mechanical resonance, (e) System with reverberation.

calculate the anechoic frequency response of a device by rejecting all reflections, or calculate the impulse response of a device without a certain mechanical resonance.

Many good acousticians who design large performing arts facilities have realized for a long time the importance of time domain analysis. The direction and early time of arrival of sound at a listener's seat is the difference between an average concert hall and a great concert hall. These two parameters are the key to describing how a listener will perceive the size and softness of the hall. Why else would these experts go to such expense in performing acoustic scale model studies?

Interpretation of Time-Frequency Data

Certain acoustic phenomena can most easily be detected in the frequency domain while others are more easily detected in the time domain. The following discussion covers some of these basic ideas.

Consider an idealized system response as shown in *Figure 1a*. Here the magnitude of the time (ETC) and the magnitude of the frequency response to a unit impulse are displayed. The system shows a fast response and little decay in the time domain and a very wide bandwidth in the frequency domain. If the system in *Figure 1a* were a loudspeaker the subjective impression would be one of a high degree of definition of transients, definition of space, and clear separation of instruments.

Now let's perform the same measurement 90 degrees off axis. The sound must now diffract around the edges of the loudspeaker cabinet and the result would look similar to *Figure 1b*. The ETC shows poor time definition due to the width (diameter) of the loudspeaker, and the frequency response correspondingly shows a more limited range. Also, because deffraction is frequency selective, the response will roll off with increasing frequency.

Some very interesting time and frequency domain measurements for microphones at various angles of incidence are shown in reference [2]. Most high quality microphones are designed to minimize this effect because the angle of incidence is not always well defined.

Figure 1c shows the introduction of a reflection. This is shown as a well-defined peak of energy delayed by the additional propagation time of the reflected path. The frequency response shows the familiar comb filter effect. Quite often, the comb filter can be mistaken for other measurement anomalies — especially if the data is plotted on a log frequency scale. Most acoustic propagation delays may be considered nondispersive; that is the shape of the pulse does not change with distance/time. This is a consequence of the compressability of a fluid medium such as air. However, there are propagation paths which are dispersive such as bending waves in solid structures. The speed of sound increases with frequency, with the higher frequency energy out racing the lower frequencies. (This gives metals their characteristic "metallic" sound.) Such a dispersive path would tend to broaden or smear the ETC.

The effect of a mechanical resonance is shown in *Figure 1d*. If this were a microphone it would cause coloration and smearing of the recorded sound. The system when excited resonates at a characteristic frequency, as is seen in the frequency domain, causing a gradual release of energy after the initial response. If the time domain energy is plotted with a logrithmic amplitude scale, such as decibels, then the exponential decay for a linear system will appear as a straight line whose slope is

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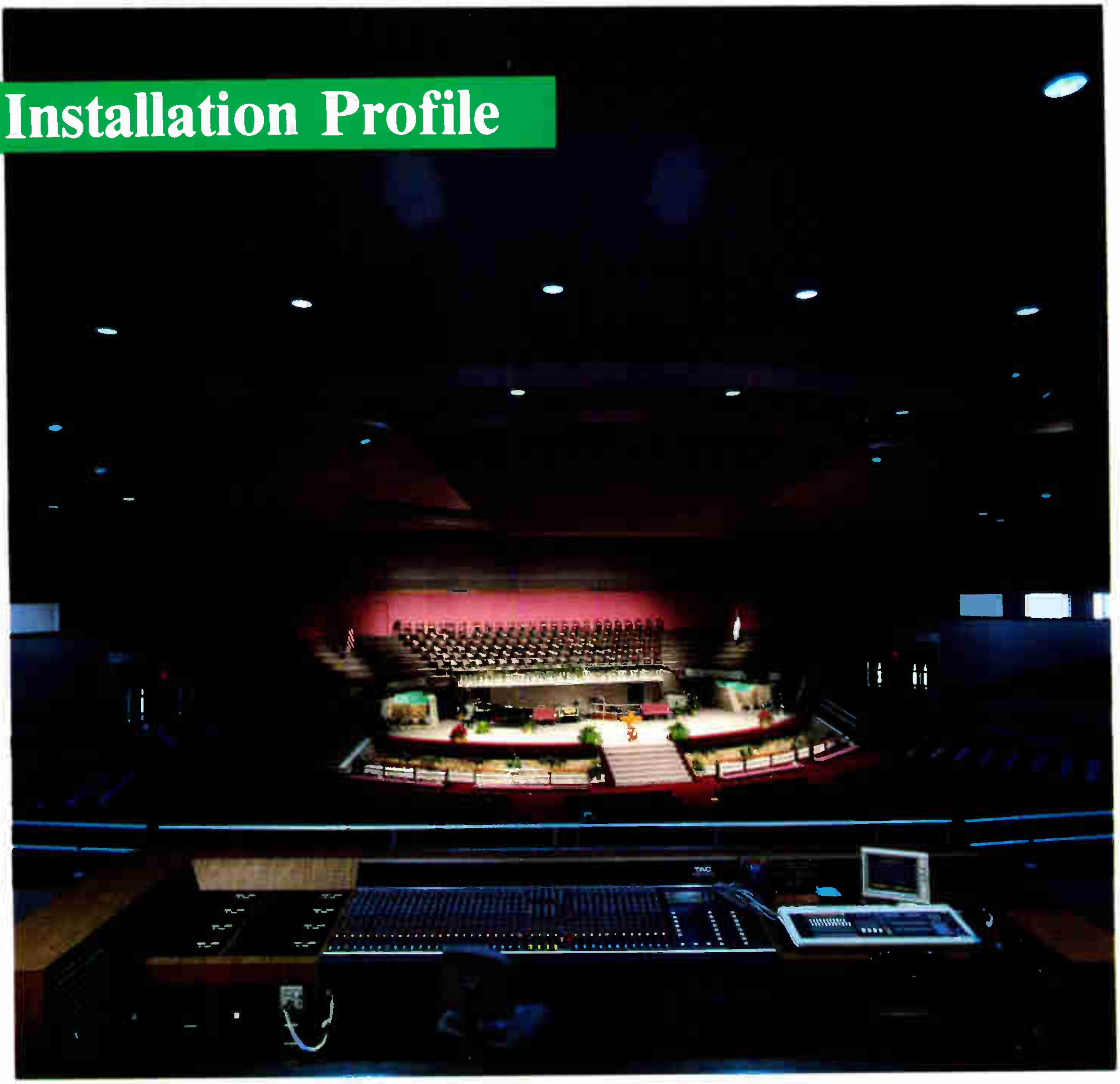


FIRST SOUTHERN BAPTIST CHURCH

by Jim Ford
Ford Audio/Video Systems Inc.

In 1984, the First Southern Baptist Church of Del City, OK, broke ground for a 7,000-seat sanctuary and 180,000-square-foot adjoining educational building. The project was completed in April of 1986 making this church one of the nation's largest.

The basic architectural design is a square building with the center line of the auditorium being rotated by 45 degrees. The resulting seating plan is fan shaped by about 110 degrees. The seating is divided into a main floor with 4,500 seats and a cantilevered balcony with 2,500 seats. The huge size of the auditorium can be visualized by the fact that a full-length football field can be placed inside the auditorium from corner to corner. The balcony front edge is

110 feet from the pulpit, and the distance from the pulpit to the most distant seat is 155 feet. The choir area seats 700 and the choir room in the educational building is larger than many church auditoriums.

The project was unique in that it was a design-build contract with the architect, builder, and owner. After the original design was completed through a series of monthly meetings during the construction phase, the project was evaluated and changed. This procedure allowed for a budget for the sound and lighting system, but did not require that any items be purchased until they were needed to meet the construction schedule. Certain items like the main mixer, microphones, recorders, and auxiliary equipment were held and new products were reviewed. This worked very well and, as usual, many products that were part of the original design were no longer available or had been replaced with newer models with improved performance and features.

Several factors in the architecture of the building were of concern. The church performs several baptisms each Sunday, consequently they had two baptistries in operation at all times. The design called for pumped water over a stone wall and down into a pool. One baptistry was on the far left of the platform, and one was on the far right. In addition to this noise problem around the front edge of the platform, which was about 90 feet long, there was a waterfall. "The River Jordan" pumped water down a five-foot wall of rocks into a pool. In preparation for a high noise level the controls for the pumps were placed at the sound console. In actual use, the waterfalls did not cause a great problem except when the overflow valve failed and the flood filled parts of the sound system conduit.

At both sides and at the rear of the auditorium, the entry foyers were atrium-type openings that connected directly to the auditorium over the rear of the balcony. This area was constructed of hard surfaces and made a nice reverb chamber. Any noise made in the foyer was audible in the balcony. Also these separate rooms provided an after ring that could be heard in the auditorium after the sound decayed. One last concern was the noise specification of several of the air vents in the balcony was not met.

The church music program is progressive and they perform contemporary Christian music on a weekly basis. They have a full-time orchestra and they present large dramas and musicals several times a year. Due to the contemporary program and the large size of the room, it was decided to design the system for a higher ratio of direct to reverberant sound than found in a church with more traditional services. The reverb time was held under two seconds. The interior wall and ceiling surfaces are Sheetrock, the floors are carpeted, and pews are padded. The room was analyzed for echoes based on the majority of sound energy being projected from the platform and the speaker cluster. The orchestra was permanently located in a recessed pit area between the choir and the main preaching platform and this worked well in controlling the sound level so that the choir could be heard over the orchestra.

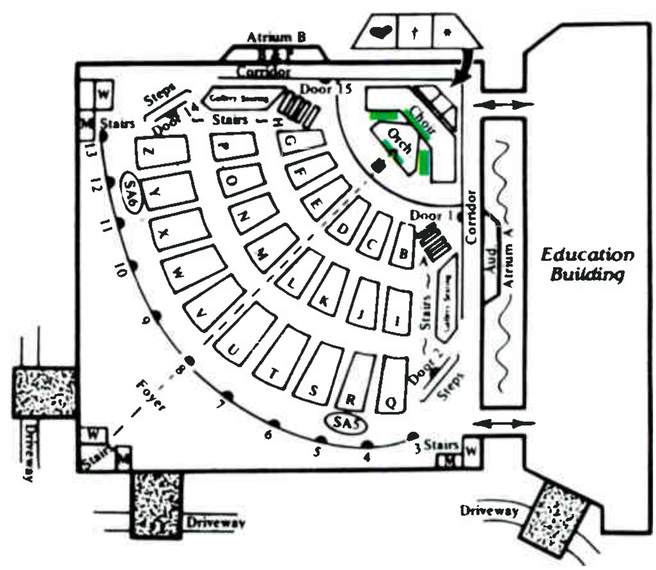
The main auditorium sound system is composed of a central cluster, an over and under balcony signal delayed system, and a front fill system. The main cluster is about 35 feet above the platform and consists of four JBL 2366 long throw horns to the balcony, five JBL 2365 medium throw horns for the rear and sides of the main floor, and three JBL 2360 short throw horns for the front of the main floor. All the high frequency drivers are JBL 2445. The bass enclosure is a tuned column with eight JBL 2225 15-inch speakers.

The front fill system was designed to cover the first three rows of pews in order to provide presence from the platform in an area where the main cluster is virtually over the head of the listeners. This system used eight JBL LE8T eight-inch full range speakers which were recessed into the front of the platform (above the River Jordan). The speakers were signal delayed and equalized to blend with the natural sound of the platform and the reinforced sound of the cluster.

The under balcony overhang was more than 30 feet and the ceiling height was only 10 feet, resulting in the majority of the under balcony being shielded from the direct sound of the cluster. The system for this area consisted of 150 JBL 8140HTWB divided into three delayed zones. The foyer speaker system at the rear of the auditorium was also connected to the third delay zone. As would be expected the area under the balcony has a shorter reverb time than the rest of the auditorium.

The over balcony system used the same speaker type and was connected to the appropriate delay zone. The purpose of the over balcony system was to maintain the sound level, because the sound projected from the long throws in the cluster was diminishing due to increasing throw distance and angle of projection. All of the balcony could see the cluster and there was not a problem of having direct sound. The problem was maintaining the desired ratio of direct to reverberant sound.

Crown amplifiers and Urei equalization are used throughout the system. Third-octave equalization was used on the main cluster and all monitor sends. All other sends in the system used octave equalization. The under and over balcony systems were protected by a Urei compressor/limiter. If the church were to bring in a contemporary Christian music group the cluster would produce as much sound level as they would need, but the available audio power and power handling capability

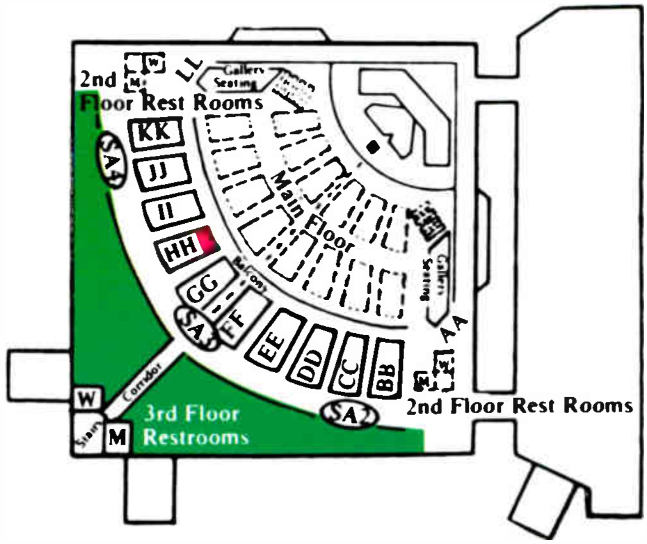


The floor plan of the main level of the Southern Baptist Church. The baptistries, each with a waterfall, are highlighted.

of the under and over balcony speakers would be exceeded. Consequently the limiter was set not to affect speech but to control the maximum level of music.

The monitor system is divided into six sections. The front platform has one send from the console and is powered by three Crown DC300 amplifiers. JBL 4602s are used for speakers. The orchestra is provided with three console sends. There is an overhead monitor system which uses two JBL 2366 long throw horns and one tuned bass enclosure with four JBL 2225 speakers. The object was to have a very directional array so that the monitor sound was projected to the orchestra and not to the main platform or choir. The second and third orchestra system is comprised of several Crown power amplifiers serving monitor jack panels around the orchestra pit walls. This allows 16 small monitors with volume controls to be used by the musicians. The choir receives two sends, one being for an overhead system and one for a rear surround system. The overhead system consists of four small clusters with one JBL 2365, 4560, and 2225 each. Once again the design goal was to provide a directional array that would keep the choir monitor sound on the choir. The second choir system used 10 JBL LE8T eight-inch speakers that are spaced around the rear of the choir in an overhang that is below the organ chambers. Due to the large size of the choir area and the height of the overhead speakers which would cause a delay, the surround speakers were designed to be used with music programs such as the choir singing with prerecorded tapes.

The balcony level—the green highlighted area is open from the foyer to the balcony. The red area is the mixing console position.



All of the monitor sends are controlled at a custom panel at the sound console. All of the monitor sends from the console go to a set of rotary switches that allow any send to be connected to any monitor system. This panel also controls via relays all of the speakers (horns, bass, and monitors) in the auditorium so that the sound man can change the coverage pattern of the sound system. The auditorium may have a group of 200 or a full house and the sound man can alter the coverage of the system easily to meet the needs. This system also allows quick verification of the operation of any horn, bass, or delayed speaker system.

The main mixer is a TAC Matchless with 36 inputs. Twelve 6 channel submixers are used to pre-mix the choir, orchestra, and music groups. There are 150 mic lines normalized through ADC pro-patch patchbays to the PA side of the system. Jensen transformers are used to split the mic lines to TV which are also through patchbays. The church has a 40 foot video truck with one-inch video tape and Ikegami cameras. Several limiters, a digital reverb, two Nakamichi cassette decks, and a reel to reel are interfaced to the main console via its line level patch-bay.

The system provides highly intelligible speech and quality music at long throw distances with excellent gain before feedback. The coverage and frequency response are smooth.

Jim Ford, president of Ford Audio-Video Systems Inc., has been involved in all aspects of audio for over 20 years. Ford holds an electrical engineering degree from the University of Oklahoma and is a member of ASCAP, AES, and ASA.

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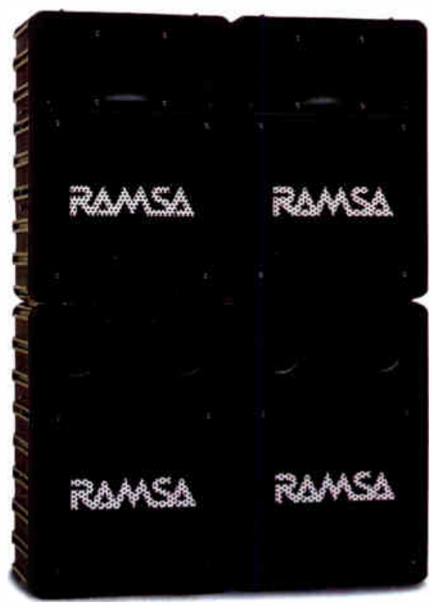

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THE ORIGINS OF MIXING FOR LIVE SOUND REINFORCEMENT

THE EVOLUTION OF THE MIXING PROCESS IS OBSCURE due to its lack of readily available documentation. With the proliferation of so many mixing consoles in today's marketplace, it is interesting to learn of the origins of this black-art. From the early days of radio and telephone and subsequently talking motion pictures; microphone techniques, the mics themselves, and all the ancillary supporting electronics were developed to supply a need—to communicate information.

During the first decade of radio, most of the industry was primarily concerned with making the medium work. Voice transmission was a miracle, therefore the primary concern. When motion pictures first acquired a voice, the industry and the public had a rough time. When asked to speak or sing, many silent-screen actors did not make the grade. As sound was accepted by the studios, recording it on the film became a standard (as opposed to a synchronized operation with sound records) and the art and illusion of the motion picture began its technical conquest.

While at that time all systems were monaural, attention was being given to natural reproduction using the control of loudness and spectral characteristics. To balance the attributes of a sound source meant the introduction of equalizers and variable-gain microphone preamps. At first, the level of a microphone preamp was changed by having a rheostat control the voltages applied to the vacuum tube. Radio

Leopold Stokowski, sitting at the control panel, and Harvey Fletcher (standing) in Constitution Hall in Washington, DC during the stereophonic long-wire transmission tests with Bell Labs in 1933.

AUDIO PERSPECTIVES

UPI/BETTMAN

BY JESSE KLAPHOLZ

broadcasts were mostly done with a single microphone for the pickup of sound at a given location—at first with carbons, and later with condenser and ribbon mics. The microphones used at this time were carbon types and the technique of "following the action" meant using a "switchboard" that would turn on and off the microphones that would be appropriate for the transmission. These techniques were especially important to minimize distortion and noise of the carbon mics. Also, the sound engineers were aware of the lack of realism when using many close-mic positions as exemplified in this statement by John Cass, a motion picture sound engineer in 1930:

> *When a number of microphones are used, the resultant blend of sound may not be said to represent any given point of audition, but is the sound which would be heard by a man with five or six very long ears, said ears extending in various directions.*

Also of concern were the poor phase relationships when combining a number of microphones in an amplifier, as carbon mics were placed close to individual sources. The technique developed was to use a switchboard that would allow an operator to follow the action and turn mics on and off, thus overcoming noise and phase distortion problems. Needless to say, these early sound technicians were frequently in heated arguments about the *sound.* Perhaps typical of the comments about early sound technicians was those made by C.W. Horn in a radio address over KDKA in Chicago on January 3, 1923:

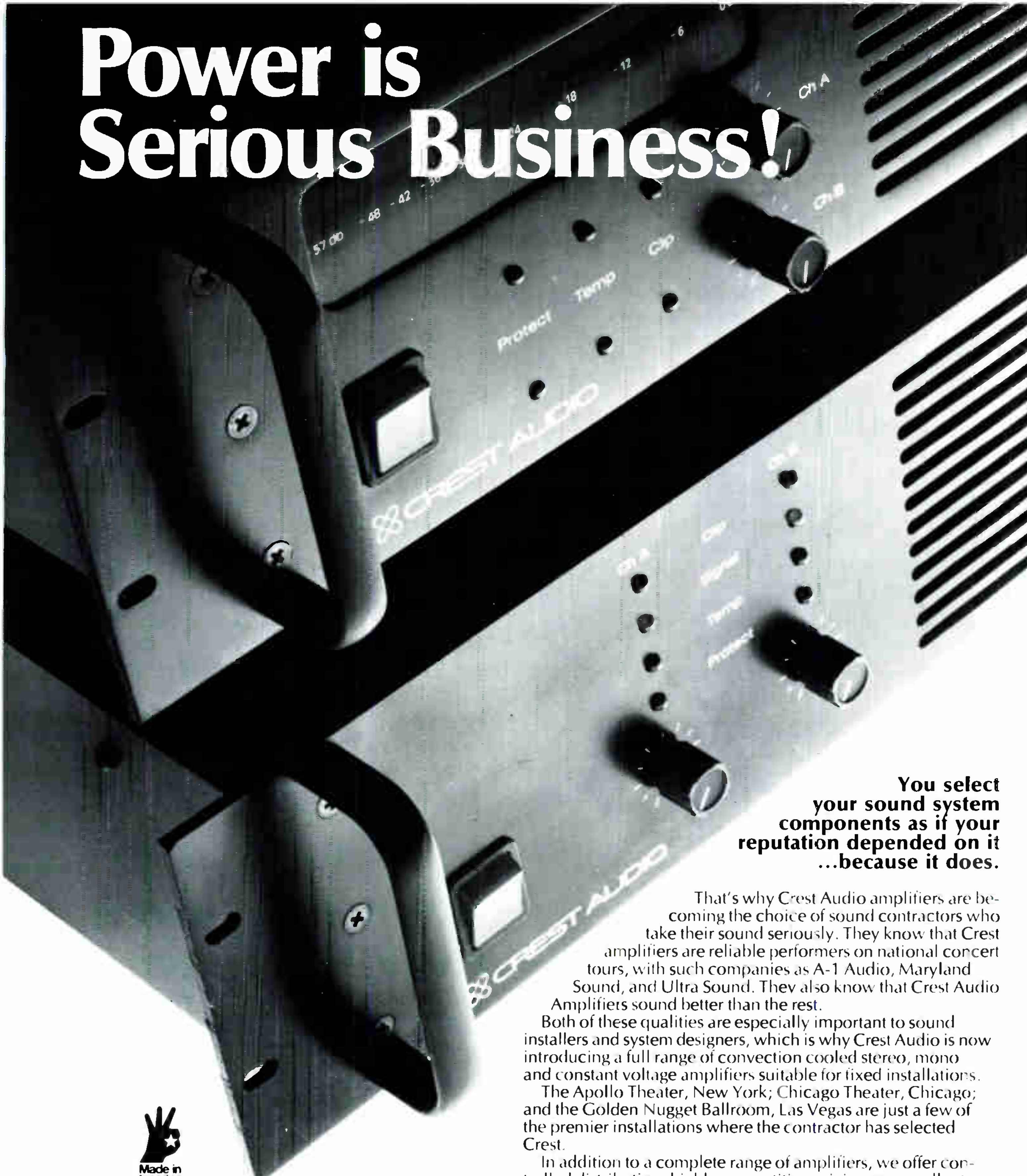
> *At Chicago our station KYW has 10 microphones scattered about the auditorium where the Chicago Civic Opera Company renders its selection. An "expert" sits in the audience with a small switchboard in his lap and cuts in the proper microphone for whatever type of performance is being offered at that instant. He uses a different [not type] microphone for a solo than he does when the orchestra is playing, and he must make the change instantly. . .Needless to say this man knows all the operas by heart.*

When the condenser mic was made available to the broadcast industry it simplified the setup to a single mic, since it had ample sensitivity and signal-to-noise ratio to place it sufficiently far from a source to pick up the presentation in proper proportion and blend. These new techniques included the importance of clearly understanding the directional patterns of microphones, the acoustical environment, and the nature of the acoustical characteristics of the source(s). All of the blending was done by the careful positioning of a *single* microphone.

Leopold Stokowski's achievements in the field of electro-acoustics can only be stated here to a limited degree. As the conductor and musical director of the Philadelphia Orchestra, he led the orchestra in a number of important firsts: the first orchestra to record electrically, the first to give commercially sponsored live symphonic broadcasts in America, the first to be featured in a motion picture, the first to transmit stereophonically, the first to perform electronic music in America, and the first to be televised live.

Stokowski's interests in electronics started with his exposure to Thadius Cahill's Telharmonium in 1906. Stokowski had a lifelong dream of a

Power is Serious Business!



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"temple of music." The headline in the Sunday, October 6, 1929 edition of the Philadelphia Inquirer marked the event:

> *"PHILADELPHIA MAKES HISTORY ON THE AIR TODAY EMPLOY NOVEL METHODS TO PUT ORCHESTRA ON AIR—Engineers Meet Problems of Broadcasting Philadelphia Symphony This Afternoon With Radio Technique Never Before Used."*

NBC sent O.B. Hanson, manager of plant operation and engineering, and a corps of engineers to the Academy of Music. They were responsible for the pickup of the orchestra and announcer. The novel method was the placement of a single condenser microphone on a high stand and focused so it would pick up the orchestra ". . .just as a pair of human ears in an audience would receive the sound waves." After the broadcasts Stokowski listened to the playbacks. He was not pleased with what he heard. He felt that the dynamic range was too compressed.

Stokowski's solution was to remove the control of level from the engineer, have a meter mounted at the podium, maintain the gain of the system at a fixed point, and control the dynamics of the orchestra himself. According to the December 15, 1929 issue of *Musical America*, "All three pick-ups [the following three broadcasts] were more perfect than anything which had gone before. . . ."

Of even further importance was an announcement Stokowski made during one of the broadcasts of his intention to spend his upcoming 12-week vacation studying radio engineering. At this time the Moore School of Electrical Engineering at the University of Pennsylvania invited Stokowski to participate in a series of experiments on musical acoustics. He worked with Professors Harry Hart, Walter Lusby, and Charles Weyl. His studies and experiments became a passion, and he was constantly experimenting with the placement of musicians, instruments, and microphones.

Harvey Fletcher had contacted conductors Koussevitzky and Toscanini soliciting the participation of their orchestras in his experiments—neither was interested. In April of 1930, Stokowski visited Bell Labs where he first met Fletcher, the director at the lab. It was at this point that Stokowski would begin a long relationship with Bell Labs in addition to his established working relationship with RCA. Perhaps, Stokowski in his infinite wisdom had both sides playing against each other, nurturing competitive research, just as the movie moguls had done before.

In April 1931, Harrold D. Arnold of Bell Labs had sent a number of books on acoustics to Stokowski, which he thanked Arnold for in a letter and added: ". . .if. . .I or the Philadelphia Orchestra can be of any service to you in any sound experiments we are always at your disposal." Subsequently, Stokowski agreed to have elaborate equipment installed in the basement of the Academy. Among the engineers at Bell Labs to work with Stokowski were Arthur Keller, Harrold D. Arnold, and Joseph P. Maxfield. Keller and Maxfield began experimenting with microphone pick-up positions with Stokowski. Large loudspeaker systems were set up in the ballroom of the Academy to audition the results of the various experiments. These experiments culminated in the development of what Fletcher called, "Auditory Perspective." Stokowski approved of the experiments, and said, "Listening binaurally gave me more sense of space. . . .I found it better in every way than monaural listening."

With the approval of Stokowski, the experiments proceeded with almost 130 recordings and several private demonstrations. The first large-scale demonstration was done with the orchestra in the ballroom and loudspeakers on the stage—Stokowski controlled the dynamics.

The first public demonstration was given in the 1933 Bell Laboratories' historical demonstration of three-channel stereophonic long-line transmission between Philadelphia and Washington, D.C. E.C. Wente and A.L. Thuras specifically developed the two-way horn-loaded loudspeakers used for this study. In those days, 10 watts of audio power was about as much as could be produced, and these systems could deliver the necessary acoustical levels with only that small amount of power. These same kinds of loudspeakers are still in use today in cinema and large scale sound reinforcement applications.

In 1979, at the age of 94, Fletcher was asked if Stokowski actually did the mix, handling the dials.

> *"Yes. He handled them. Nobody but a great musician could do it. . . He could, by turning knobs or touching buttons, make the violins come way up or any part of the orchestra. He loved that when he found what he could do. . . . He could correct things that he saw ought to be corrected!"*

Linton Martin of the *Philadelphia Inquirer* wrote: "For those alive and alert to the significance of the episode, it took rank as an epochal event in the history of musical performances. In fact, Stokowski, in his zeal to produce a huge climax, accidently twisted off one of the mixer's knobs."

Later, when Stokowski was conducting in Los Angeles in 1935, he arranged for another demonstration in the Hollywood Bowl. After the concert, he recommended the shell be torn down and replaced with an electro-acoustical shell! In the late 1930s, Bell Labs was working on recording sound directly on film. Fletcher brought the new film recorders to the academy in Philadelphia and recorded the orchestra on three tracks. In making the final transfer of the tapes, Stokowski manipulated the dials to "enhance" the final version. It was that performance that was demonstrated in Carnegie Hall in 1940. Fletcher explained that as Stokowski listened, he "made volume and tonal changes by electric controls; and simultaneously a new stereophonic record was made of the music and thus 'enhanced.' "

Stokowski was certainly recognized as a giant in his day by the broadcast/recording/film companies, as well as the technicians and engineers. On November 2, 1931, he received a medal from CBS for his "distinguished contribution to radio art," which was presented by William Paley. On December 9, 1931, Stokowski was invited to speak at a joint meeting of The Institute of Radio Engineers (IRE, now called the IEEE) and The Society of Motion Picture Engineers (now called SMPTE). At the engineers club, after covering the technical aspects of his experiments with Bell Labs, he offered this perspective: "The limitations of music are becoming less and less. . . I believe the composer of the future will create his harmonies directly in tone by

(continued on page 43)

(continued from page 20)

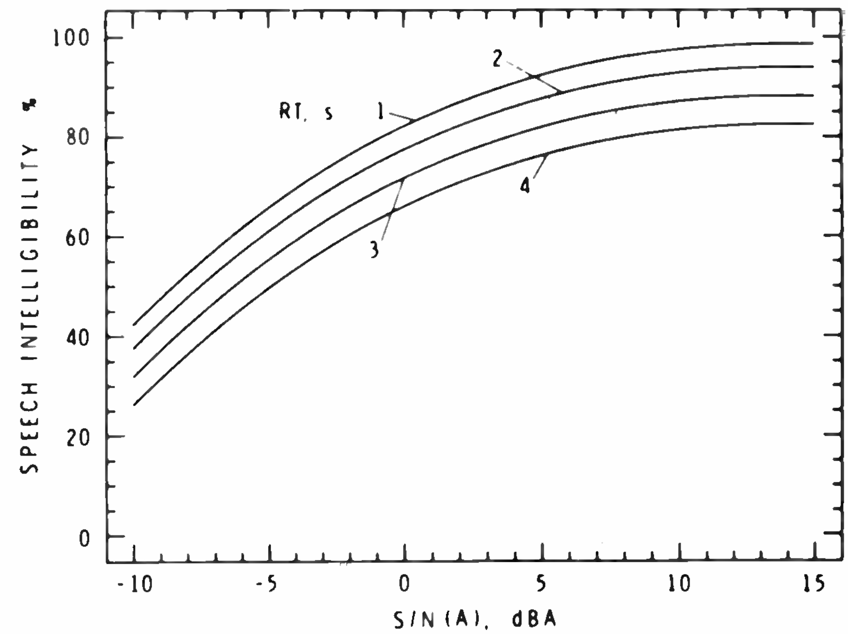
4. view of talker/sight of listener .17
5. speed of talker 1.00 or >
6. hearing of listener 1.00 or >
7. motivation of listener 1.00 or >
8. distance to microphone .40 or >
9. speaker orientation .40

The application of perceptual psychology theory would begin to address some of these issues, via such tools as signal detection theory. While this listing is by no means complete, it suggests that we must develop a much more comprehensive definition of the problem of sound system design, so that the final design is not overcome with unconsidered failures.

Recent Developments

At least three new issues will be raised as this discussion progresses. First, Peutz has now proposed another intelligibility metric entitled the "Information Index," which is a broader attempt to characterized intelligibility. Second, Richard Heyser is now beginning to consider the application of sound intensity technology to the simulation of binaural auditory process that more accurately simulates the process of hearing. He has recently noted that differences in so-called quality that cannot be discriminated via the single microphone process can be simulated by two position sound intensity measurement. Finally, J.S. Bradley of the National Research Council of Canada has supported the concept of using a weighted signal-to-noise ratio and 1,000 Hz reverberation time to directly predict speech intelligibility changes, in accordance with the graph in *Figure A*. He claims that this method, among others he has tested, has proven reasonably reliable, thus bringing us full circle back to simpler evaluation.

Figure A



Conclusions

While the complexity of current measurement suggests that the average practitioner is in the process of being overcome with technology, the current results of sound system design are indicative of another conclusion. In fact, most sound system designers are having many successes with little use of technology. Secondly, many users of the more advanced testing concepts often forget that the test process is only verification of experience that has been judged subjectively. If the instrument disagrees with the ear, the instrument is generally in error, either in data collection or in the assumptions as to data analysis. Thus, instrumentation only quantifies what is already known subjectively.

Being a user of all of the technologies discussed, I would recommend the following:

(1) If the user is technically based and is measuring complex sound systems in large rooms on a regular basis, consider purchasing a TDS system, such as the TEF 12 computer. Assume that it is a very difficult analyzer to use and will require about six months of use to be moderately proficient. Also assume that untrained staff members should *not* attempt to use this device.

(2) If the user has requirements as above, but either the main user or his staff is not technically inclined, purchase a RASTI system for analysis/verification. Also, consider a RASTI system for quick problem evaluation in addition to the TEF machine, as I do.

(3) For both categories above and for users that do not have requirements for large-scale testing, use a conversational word test tape, similar to the test format described in the PB word test to verify performance. Additionally, associate with someone who is proficient for large-scale testing needs.

(4) Designers either using TEF, RASTI, or word score tests should consider the Bradley "Best Fit Curves" as a design and testing guideline in their work to determine their own view of the validity of this procedure.

(5) All designers of audio systems should note that most of the issues of reduced intelligibility become complex in environments that exhibit one or more of these characteristics:

a. large spaces—250 or more audience
b. reverberant spaces—1.5 sec or greater
c. oddly shaped spaces—domes, barrel, vaults, etc.

(Most discussions of sound system intelligibility are intended to deal with difficult cases which make up a significant minority of the contractor's work.)

Footnotes

[1]Jerry A. Fodor, "Functional Explanation in Psychology," *Readings in the Philosophy of the Social Sciences*, Brodbeck (New York: The Macmillan Company, 1968).

[2]Rein Pern, "Acoustical Variables in Open Planning," *JASA*, September, 1970.

References

ANSI. *American National Standards*

Method for the Calculation of the Articulation Index. ANSI: S 3.5 (1969).

Bradley, J.S. "Auditorium Acoustic Measures from Pistol Shots." *JASA*, July, 1986.

Bradley, J.S. "Predictors of Speech Intelligibility in Rooms." *JASA*, September, 1986.

Bradley, J.S. "Speech Intelligibility Studies in Classrooms." *JASA*, September, 1986.

Brodbeck. *Readings in the Philosophy of the Social Sciences*. New York: The Macmillan Company, 1968.

Carterette and Friedman. *Perceptual Ecology, Handbook of Perception*, Vol. X Academic Press, 1978.

Harris, Cyril M. *Handbook of Noise Control*. New York: McGraw-Hill, 1979.

Hawley, Mones E. "Speech Intelligibility and Speaker Recognition." *Benchmark Papers in Acoustics*. Dowdon, Hutchinson, and Ross, 1979.

Heyser. "Discussion at Syn-Aud-Con Intelligibility Seminar," Chicago.

Houtgast and Steeneken. "A Tool for Evaluating Auditoria." *Bruel & Kjaer Technical Review #3*. 1985.

Houtgast and Steeneken. "The Modulation Transfer Function in Room Acoustics." *Bruel & Kjaer Technical Review #3*. 1985.

Lochner J.P.A. and Burger, J.F. "The Influence of Refections on Auditorium Acoustics." *J. Sound Vibration*. January, 1964.

Peutz, V.M.A. Syn-Aud-Con handout of the Information Index.

Schubert, Earl D. "Psychological Acoustics." *Benchmark Papers in America*. Dowdon, Hutchinson and Ross, 1979.

Toole, Floyd E. "Loudspeaker Measurements and Their Relationship to Listener Preferences: Part One." *JAES*, April, 1986.

Toole, Floyd E. "Loudspeaker Measurements and Their Relationship to Listener Preferences: Part Two." *JAES*, April, 1986.

Toole, Floyd E. "Subjective Measurements of Loudspeaker Sound Quality and Listener Performance." *JAES*, January, 1985.

USA Standard for the Measurement of Monosyllabic Word Intelligibility. (ANSI 3.2, 1960).

Steven J. Orfield founded Orfield Associates Inc. 15 years ago. The company specializes in lighting and acoustics as it relates to architecture and has done work for the Bell System and IBM. Orfield is a member of NASI, IES, and AES.

THE OXMOOR DCA-2 CONTROL ATTENUATOR AND RC-16 REMOTE CONTROL

by Jesse Klapholz & Richard Feld

The Oxmoor DCA-2 is a two-channel remote-controlled attenuator system. This unit is rather unique in that it is the first unit to be offered that uses optical shaft encoding to indicate a position of a user control. With units of this type there are two things to consider: its audio quality and its functions as a systems controller. The system consists of a single-rack-mount, two-channel audio attenuator and digital/logic master unit; plus any combination of switches, relays, and RC-16 remote controls.

The DCA-2's audio inputs are XLR-type connectors and are electronically balanced. The outputs are also XLR-type but are unbalanced and are capable of driving any load of 600 ohms or higher; optional output balancing transformers are available. The front-panel has two recessed screwdriver-adjustable trim-pots that control the gain of either channel over a ± 15 dB range. The rated maximum output is + 18 dBm.

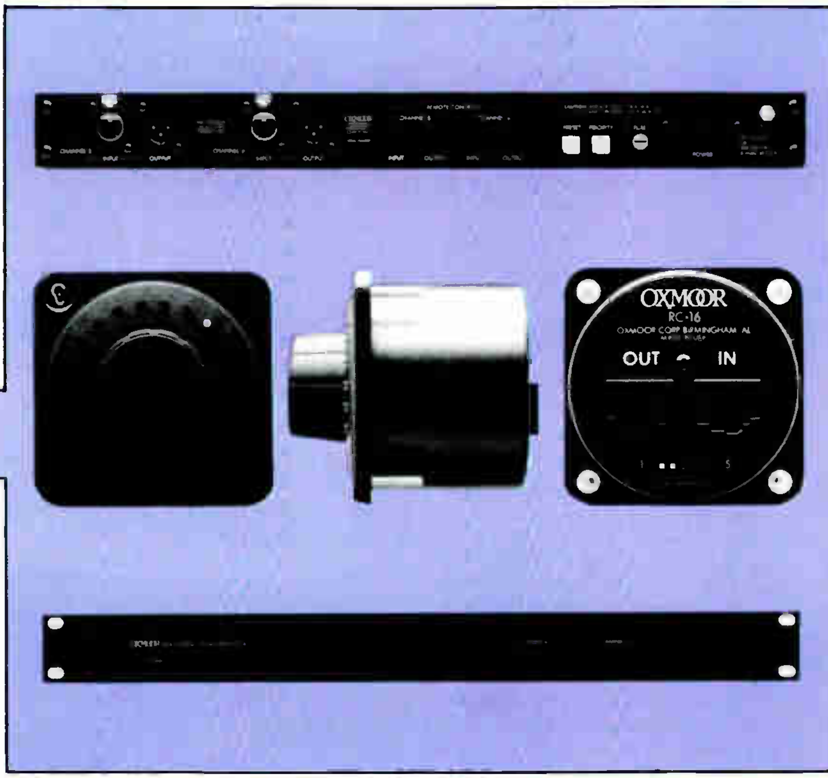
The DCA-2 has an input and output port, via six-pin modular-type jacks, on each channel. The input port receives data and/or ground closures from the remotes or any external switch. The output port loops these signals through to another channel for parallel tracking, such as for stereo. The remotes are connected to the DCA-2 with standard four-conductor telephone cable for the attenuator functions, and with six-conductor wire for both the attenuator and priority/preset functions. With standard wire, the furthest remote can be hooked up 2,000 feet from the master; with heavier gauge wire even further distances can be accommodated.

There are two preset level modes featured in the system. They both have 15 steps of attenuation in 3 dB increments and an OFF position. The *preset* is the amount of attenuation in a channel upon AC power-up of the unit or when a dry closure to logic ground is made. The priority mode is engaged only when a dry closure to logic ground is made. The preset and priority modes can be engaged by pushbuttons, key-switches, transistors, relays, and can even be easily interfaced with TTL logic signals.

The audio quality of the DCA-2 will stand up in any professional system. The circuitry is kept to a minimum and uses high-performance/low noise op-amp chips. Since the unit is an at-

Specifications:

	MANUFACTURER'S	LAB TEST'S
Frequency Response	+0dB @4Hz, −0.3dB @60kHz (@ +4dBm, maximum gain)	+0dB @20Hz & 20 kHz, −1dB @50 kHz (@0dBm, unity gain)
Hum & Noise unweighted	−80dB (ref.0dBm output @ unity gain)	−82dBm (ref. 0dBm output @ unity gain) input terminated into 600ohms
Distortion THD	0.01% ref. +4dBm, 20-20kHz	OdBm input, trim set at unity gain 100Hz 1kHz 10kHz <.003% <.004% <.009% OdPm input, trim set at −10dB <.005% <.005% <.007% OdBm input, trim set at −15dB <.006% <.007% .007%
Distortion IMD	0.01% ref. +4dBm, 20-20kHz	.004% w/0dBm input, unweighted .003% w/−10dBm input, unweighted
Crosstalk	−90dB, 20-20kHz (input terminated w/600ohms, unity gain, 2nd channel at full output)	−92dB, 20-20kHz (input terminated w/ 600 ohms, unity gain, 2nd channel at full output)
Maximum Input Level	+20dBu	+21.5dBm, 20-20kHz
Common Mode Rejection	50dB (20-20kHz)	>52dB (20-20kHz)
Max Output @rated THD	+18dBm	+22dBm
Gain Range	−15dB to +15dB	−15.5dB to +15.6dB
S/N	n/a	104dB @600ohms (20-20kHz)



tenuator, it will probably never be used as a gain-block, even though 15 dB of gain is available. In fact, in most cases the gain of the unit will be trimmed back. As in many units today, we found no real need for the additional cost of output balancing transformers. The whole idea of using this unit is to keep audio lines short and within the equipment rack.

When we put the DCA-2 on the testbench using a Sound Technology 1710 distortion analyzer, the manufacturers specifications were either confirmed or found to be on the conservative side. We found the distortion and noise to be low, and that there was more than adequate headroom and output. When we overloaded the input stage (and this took a +21.5 dBm signal) the clipping was smooth and symmetrical. When the proper input level was not exceeded, we could not clip the output. A table of the manufacturers specs and what we measured is shown in *Figure 1*.

The attenuator accounts for the audio portion of the story. On the other side, the control section uses an all digital logic scheme. The RC-16 remote control is based on a rotating shaft whose movements are precisely translated into a string of digital pulses. It feels like a high-quality detented pot, but it can be turned completely around with no mechanical stops. Instead of numbers around the dial, the remote has an array of LEDs that indicate the position of the attenuator in the master unit. All of the remotes on a given channel track each other in the display of level. Oxmoor calls this the *virtual pointer*.

The attenuator precisely tracks the movement of the remote—the faster the movement, the faster the change in level. On the back of the remote are an *In* and *Out* port for the logic signals and a five-pin connector for lockout, preset, and priority mode selectors. Also, a remote may be moved between any two circuits with a four-pole switch, or between more circuits with a rotary-selector switch if multiple DCA-2 units are used.

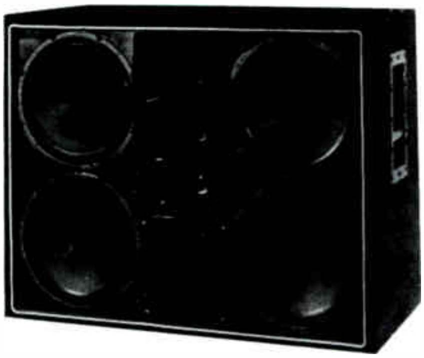
Most noticeable is the absence of a power switch. Thanks guys, you listened. We do not need to fiddle with power switches that inevitably introduce pops into the audio chain and break at the most inopportune times. In addition to the slick look of the device and the glowing LED to indicate volume levels, this system promises to solve several problems. Audio signals or precise control voltages are not being run; instead, the logic signals may be run on inexpensive wire or existing phone lines, for example, and may be run with any data or phone cables. One possiblity is to have dedicated phone jacks thoughout a complex. A technician may plug in his hand-held RC-16, set a level, *disconnect* the remote, and the system will stay at that level until changed or powered

(continued on page 49)

GENERAL SPECIFICATIONS:

Digital Attenuator Control Range	0 to −43.5dB in 29 steps plus a 90dB full attenuation ("kill") step
Tracking Accuracy	Within +/−0.1dB throughout attenuator range
Preset & Priority Range	15 steps (of 3dB each) plus "kill"
Overall Dimensions	1.75"H x 19"W x 7"D (DCA-2) 2.25"H x 2.25"W x.8" above panel; 2.1"dia.x 1.6"D (RC-16)
Weight	6.9lbs. (DCA-2) .25lbs. (RC-16)
Price	DCA-2/$560 pro net RC-16/$160 pro net
Manufacturer	Oxmoor Corp., Birmingham, AL

PRODUCTS IN REVIEW



COMMUNITY LIGHT & SOUND'S CS70 LOUDSPEAKER SYSTEM

Community Light & Sound has introduced the CS70 loudspeaker system which utilizes four specially-designed 12-inch low frequency drivers, a pair of midrange drivers with two-inch throats, and a Focused Array™ high frequency section to serve up a smooth, flat response from 45 Hz to 18,000 Hz. With a broad-band sensitivity of 105 dB (at 1 watt, 1 meter), the CS70 is capable of producing continuous SPL levels in excess of 130 dB at 1 meter, and peak levels well above 140 dB, which provides the dynamic headroom necessary for full-fidelity reproduction of live program material. The three-way, passively-crossed CS70 will handle input levels of 600 watts rms/1,500 watts program.

Designed to withstand the rigors of life on the road, the CS70 is housed in a well-braced enclosure made from high-density particle board covered with black carpet. A fabric-covered steel-mesh removable grille, recessed steel bar handles, and fuseless internal protection from excessive input signals enhance overall performance and reliability. The CS70 has a suggested retail price of $849.

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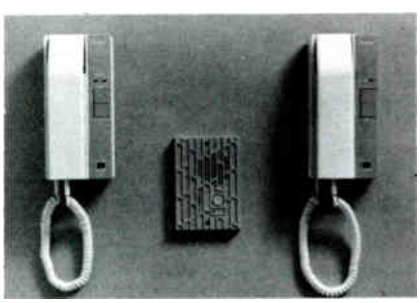

AIPHONE'S NEW IC SERIES TO REPLACE IBG SYSTEM

Aiphone Corporation has designed a door entry security intercom for residences and small businesses to replace its IBG system.

The IC Series, which interfaces with Aiphone's MB Video security intercom system to provide voice communication, can accommodate up to three inside room stations and two outside door stations. Inside stations can be used to call from room to room.

The IC Series has a more compact, modern design than the IBG and features a slim handset that barely protrudes from the wall. The system also has two and four stroke chimes that let residents know which door station is being used.

The system operates on either 24 V DC or on 12-16 V AC power. If used with Aiphone's new AC 16 V transformer, the PT-1620. The system can also activate a door release.

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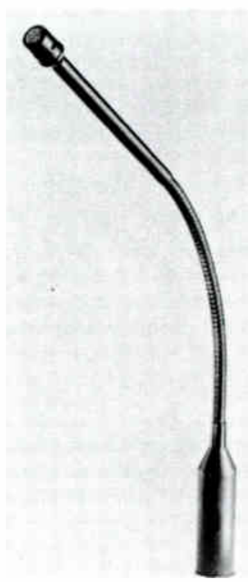

SENNHEISER'S MKE 42PU CONFERENCE LECTERN MIC

Sennheiser Electronic Corporation has introduced the MKE 42PU, a stylish gooseneck lectern microphone designed for podium and conference applications. Employing a quality miniature condenser element, the MKE 42PU has a wide cardioid pattern which, according to Product Manager Anthony Cafiero is "specifically designed to minimize the low frequency proximity effect often caused by users who are inexperienced in proper microphone technique." The wide cardioid pattern ensures against premature feedback while allowing the speaker freedom of head and body movement during lectures and speeches. A full frequency response facilitates accurate vocal reproduction.

The MKE 42PU microphone's unobtrusive design prevents the all too common "mic fright" most speakers confront when walking up to a lectern or podium. The pencil thin gooseneck is jacketed with a four-inch cylinder just below the microphone element to prevent gooseneck kinks by limiting the amount of flexure. The MKE 42PU requires no external preamplifier or battery supply as a 48 volt phantom powering adapter is integral to the base of the microphone.

Circle 18 on Reader Response Card

TOA ANNOUNCES TC HORN SPEAKER SERIES

Toa Electronics has announced a new generation of TC Series horn speakers which are available with dual 25/75 volt transformers, or at 8 ohm voice coil impedances; and power ranges of 10, 15, or 30 watts. The Toa horns, with stainless steel mounting hardware, feature elliptical, aluminum bells that are chemically treated to withstand severe weather conditions and corrosive environments.

The 10 watt, TC-101 (8 ohm), and TC-101TA (25/75V) are both available with standard "U" mounting brackets. The 15 watt, TC-151 (8 ohm) and the 30 watt, TC-301 (8 ohm) also have standard "U" mounts. Swivel mounts are available on models with internal, dual 25/75V transformers: TC-151TA (15 watts) and the TC-301TA (30 watts). These universal swivel mounts are designed for positioning horns in most any direction, and feature the ability to be strap mounted to beams or poles. This mount also has a fixture for mounting directly to threaded halfinch conduit. A 24-inch jacketed pigtail is supplied for minimum installation time; and the "wattage" selector switch is screw driver adjustable without dismantling any part of the horn.

Circle 19 on Reader Response Card

DUKANE'S POWER AMPS FOR PA & BACKGROUND MUSIC

The Communications Systems Division of Dukane Corporation has announced a new line of power amplifiers for public address, background music, and sound management applications in schools, churches, stores, hospitals, factories, stadiums, offices, hotels, and other facilities.

The new amplifiers feature broad frequency response and low distortion of less than .5 percent over full bandwidth.

Dukane's new power amplifiers, Models 1A3060 and 1A3125, have rated capacities of 60 and 125 watts continuous (RMS) power. Models 1B3060 and 1B3125 include circuitry 24 VDC battery backup.

Equipped with electronic protection and thermal circuit breaker, Models 1A3060/1B3125 are designed to prevent damage under overload or short circuit conditions.

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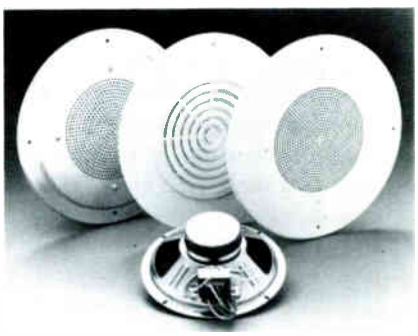

3M SPEAKERS WITH MAGNETS FOR EFFICIENCY

3M's new line of competitively priced eight-inch standard and coaxial speakers have 10-ounce magnets for increased sound system efficiency. A whizzer cone helps extend the speaker response to 70 to 15,000 Hz.

The coaxial model has a solid-state piezo ceramic tweeter, center mounted without a structural bridge, to provide crisp high fequencies without blocking the low frequency sound from the woofer. Frequency response is 70 to 20,000 Hz.

The 70-volt speaker transformers have color-coded, stripped and tinned leads for easy installation.

Speakers are available with transformers or assembled to steel, plastic, and aluminum grilles for ceiling installations. Speakers are also available in Slimline and Slantline wall cabinets with walnut-grained vinyl finish. Suggested end user list prices for the standard and coaxial speakers are $13.90 and $21 respectively, with volume discounts available.

Circle 21 on Reader Response Card

ROSS INTRODUCES 12 BAND PARAGRAPHIC EQ

The new Ross R12SP is a graphic EQ which is a combination of a standard graphic and a parametric. The Ross R12SP features a stereo 12 band EQ and each band has three selectable quarter octave center frequencies allowing the user to select the frequency area desired for optimum performance and optimum flat response setting in room acoustics. The R12SP has been designed using ultra low noise Op-Amps and a highly regulated power supply, and it has a flat frequency of 10 Hz to 65 kHz and a harmonic THD of .02 percent and an IMD of .02 percent and a signal-to-noise ratio of greater than 100 dB.

The Ross R12SP Paragraph EQ retails for $319.95. Ross also makes a two-thirds octave stereo 15 band rackmount EQ and a Mono one-third octave rackmount EQ, both for $199.95 suggested retail list price.

Circle 22 on Reader Response Card

UNEX OFFERS ASSISTIVE LISTENING DEVICE

UNEX, which has acquired the In-

(continued on page 41)

PRODUCTS IN REVIEW

a closer look

by gary d. davis

Computerized Equalization From MicroAudio

MicroAudio has introduced the Model 2800 computerized equalization system.

The Model 2800 combines the functions of a third-octave real time analyzer with a digitally controlled third-octave graphic equalizer and adds a computer for measurement and control functions. Under the control of the onboard computer, the gain settings on each of 28 ISO-centered filtering bands can be stored and recalled from 16 memory locations. According to MicroAudio, specific requirements for EQ can be recalled on demand by the "push of a button."

The Model 2800 incorporates a digitally controlled RTA which can display the spectrum of an audio source on the 28 band LED matrix to locate acoustic anomalies. Through the use of an internal pink-noise generator and sensing microphone (48V phantom power is self-contained), the 2800 will perform computer-controlled automatic equalization and EQ your system to any memory setting. The Model 2800 also possesses the ability of instantaneous automatic EQ and RTA curve averaging of up to eight curves and can be weighted as much as 7:1. Sensitivity of EQ or RTA settings can be changed from 3 dB to 1 dB by the push of a button.

For the sound contractor, the Model 2800 acts as the *master* computer to down-load into and EQ POD, any prescribed room equalization curve. The EQ POD is a blank-panelled, third-octave *slave* equalizer for permanent sound system installations and cannot be adjusted by unauthorized persons without the Model 2800. The EQ PODs 1.0 and 1.1 each remember a single equalization curve. The EQ POD 1.2 stores up to eight equalization curves, recallable *only* with a three digit access code.

Comments: To my recollection, dbx/BSR was the first to introduce an automatic graphic equalizer/real time analyzer. Though that unit was designed for the consumer market, it led to a professional version. MicroAudio has taken the concept a step further, with a lot of memory for RTA settings and EQ curves.

There are a number of potential uses for the Model 2800, but the primary use is to *tune* sound reinforcement systems. What intrigues me about the unit is its ability to measure and record real time analytical curves for up to eight different locations. These curves can then be averaged, giving more priority to some locations than others, using the *weighting* feature of the unit. This intelligent averaging should greatly simplify the task of gathering system response data and then applying it to do a reasonable job of tuning the system. Since up to 16 EQ settings also may be stored, it is possible to try "what if" solutions to a given system, then instantly switch back and forth to compare the results. Doing this manually is almost impossible since one's ear forgets the sound of the previous settings by the time all the EQ sliders are reset (and you can almost never reset them by hand as accurately as this unit can by computer control).

MicroAudio has apparently given the contractor an excellent tool in the Model 2800. Given the "PODs," which are basically remotely programmable EQ modules, a contractor can measure and set up a given system, then install that same EQ capability at about one-quarter the cost of the equipment that was required to do the original setup.

Incidentally, Gene Rimkeit, president of MicroAudio, said the units have a noise floor of around -90 dBm, THD below 0.01 percent, and elec-

(continued on page 43)

[redacted]

PRODUCT
(continued from page 39)

frared Professional Sound System (IPSS) from its sister company Controlonics, is now offering the system to the contracting industry. The IPSS is a wide-area lightwave listening system that allows the hearing impaired to hear performances in public assembly areas, such as churches, theaters, and schools. The system will produce up to 112 dB sound pressure level —enough sound for those with moderate to severe hearing loss.

The IPSS employs a transmitter that uses invisible light waves to send speech or music to individual, lightweight, wireless receivers. The transmitter, which covers 4,000 square feet, consists of a power supply, LED array, and transformer all in one unit. Slave units connect in series with the master transmitter to cover larger areas. The three ounce receivers produce high fidelity audio at personally adjustable levels and operate over 100 hours on two AAA (penlight) batteries.

Circle 23 on Reader Response Card

PREPROGRAMMED COURTESY TELEPHONE

Teledial Devices, Inc. has introduced an economical courtesy telephone for use in public installations such as hotels, airports, retail outlets, and hospitals.

Designated TDP-1000-18, the new phone lets users automatically speed-dial any one of 12 preprogrammed numbers at a single touch, while preventing dialing to other unauthorized destinations.

The fully modular TDP-1000-18 provides for speed-dial numbers of as many as 22 digits each. Numbers are easily programmable and can include pauses, hookflash and * and # frequencies as well as numerals. The set is PBX/Centrex compatible. The unit can be used on both rotary and tone phone lines and is mountable on either desk or wall.

The phone is line powered with a lithium battery to protect the memory in case of line failure.

Circle 24 on Reader Response Card

GAUSS COAXIAL SPEAKER FOR CLUBS & DISCOS

Cetec Gauss has announced a new 12-inch coaxial loudspeaker which will handle 400 watts music power (200 WRMS). The new 3285 follows the same design parameters of the company's 15-inch coaxial. A criterion for the computer-designed coax was to eliminate the need for costly time delay networks. The two drivers are well within the Blauert and Laws' criteria for perceivable time delay.

The high-frequency horn does not extend beyond the frame allowing flush installation in enclosures or ceiling installations. Triple spider construction assures that the voice coils are kept in the gap even at high power operation.

Featuring high sensitivity and point-source effect, the 3285 can be used for stage monitoring, discos, and small club PA systems.

The 3285 delivers 70 Hz to 13.5 kHz and has a sensitivity of 99 dB, 1 watt, 1 meter. Suggested list price is $635 ($840 with frequency dividing network).

Circle 25 on Reader Response Card

PHILIPS' MINIATURE CCD OBSERVATION CAMERA

Philips has introduced a compact LDH 0600 solid state remote observation camera.

The camera features an image sensor which eliminates microphony and lag. And because the LDH 0600 is not affected by shock or vibration, its insensitiviy to magnetic fields, and the quality of pictures it relays, the camera will also have a role to play in robotics, image processing and automation systems. The dimensions and weight, 400g, makes concealment easy. The new solid state camera's dimensions are 66 x 111 x 69 mm.

Circle 12 on Reader Response Card

INTERPRETING

(continued from page 23)

proportional to the damping. Henning Moller suggests that mechanical resonances in loudspeakers are probably creating the most audible effect in today's audio systems 3.

A similar effect is seen from room reverberation in *Figure 1e.* Except that the density of resonances may be so great that individual resonances cannot be identified in the frequency curve. If there is strong frequency-selective absorption in the room or if two or more rooms are coupled by a small opening a large group of resonances may decay at a different rate. In this case, a change in slope may be observed in the ETC when one group of resonances decays leaving only the other group of resonances remaining.

Measurement Considerations

Considering the utility of both the frequency and time domain measurement tools, it's not surprising to see more of this data presented in the literature. Unfortunately, there are several ways that time-frequency measurements can be made and it is important for authors to give enough details about the measurement to allow proper interpretaion of the data.

One of the more popular measurement methods used today is based on TDS. Sufficient information should be provided to define the frequency and time resolution. The full scale frequency is dependent on the sweep rate and the total sweep time: $F = S{*}Ts$. This defines the time resolution: $\Delta t = 1/F$. The full scale time value is $T = B/S$, where B is the bandwidth of the swept filter. (Another way to look at it is that 1/B may be viewed as a gating window.) The frequency resolution is then: $\Delta f = 1/T$.

Time-frequency measurements may also be made using a time-gating system. A spectrum analyzer or computer may also be used with what can be called *scan analysis*. In the gating method a family of frequency response curves is generated by successive measurements with the gated window (width T) delayed in increasing steps. The gated window may overlap prior or successive measurements, depending on the delay increment. The frequency resolution is $\Delta f = 1/T$. However, the time resolution is T. The delay maybe smaller; but no additional information is obtained. This is why gating methods have limited applications. There is a direct trade-off between time resolution and frequency resolution because the measurement is being done in real time. TDS does not suffer from this trade-off because the measurement is essentially not in real time.

Scan analysis is the same as analog gating, except that it is done digitally with a micro-computer-type spectrum analyzer. A large record of time data is digitized and loaded into a buffer. The time data is then processed in small blocks of data (same as a gated time window) to calculate a family of FFTs, each FFT corresponding to a different time delay. The blocks of time data are usually Hanning weighted and overlapped. This means that each frequency curve is not independent of the other; but instead, contains part of the same information as adjacent spectra. Scan analysis has one advantage over analog gated systems—successive measurements are not required; thus, it is well-suited for analyzing transient events like an impulse response.

Summary

Many of the advanced tools discussed here are not within the budget of most everyone in the audio business. Scan analysis systems are now readily available at low cost. These systems generally consist of an analog-to-digital card and software for a personal computer. TDS systems, although higher in price, are also gaining widespread use. We are now at the threshold of a data explosion. It is now easier than ever to produce large amounts of data, very quickly and at a relatively low cost.

Unfortunately, a lot of the data is confusing and, at worst, not valid. With scan analysis, there is a potential for misinterpretation. Results using these systems can be formated into three-dimensional plots and the amplitudes can be scaled in units of energy. However, these results are not equivalent to TDS data—the time and frequency resolution is usually different.

Furthermore, the data can be smoothed to any degree desired by using overlap processing and weighting functions. All of these measurement parameters need to be provided along with the data.

If only one lesson is learned here, it is that the three-dimensional plots of energy versus time and frequency are overused. This article illustrates how common audio problems are best detected and diagnosed in either the time or the frequency domains—not both domains.

References

[1] Bruce Bartlett, "TEF: Isolating Sound for Measurement," *Sound & Communications*, June 1986, pp. 28-31.

[2] Phillip White, "Evaluation of Studio Microphone Performance Using Time Delay Spectrometer Techniques," *Bruel & Kjaer Instruments Application Note BO-0075-11*, presented at the AES Convention 1982.

[3] Henning Moller, "Multi-dimensional Audio," *Bruel & Kjaer Instruments Application Note 17-206*, presented at the AES Convention 1977.

E. Curtis Eichelberger is an applications engineer for Bruel and Kjaer and holds a B.S. in Electrical Engineering and an M.S. in Acoustics from Pennsylvania State University. Eichelberger is a member of the Acoustical Society of America, the Institute of Noise Control Engineers, and the American Society of Mechanical Engineers.

AD INDEX

MIXING

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means of electrical-musical instruments which will record his idea exactly."

On May 2, 1932, in New York, Stokowski was invited to address a meeting of The Acoustical Society. His speech, given extemporaneously, was published in the July 1932 issue of The Journal of the ASA under the title "New Horizons in Music." And for his work on stereophonic sound both he and Fletcher were given medals by The Audio Engineering Society. As an interesting aside, Stokowski had also worked with Alan Blumlein in England and Dr. Braunmuhl in Berlin experimenting with stereo and binaural recording.

While it is true that microphone mixers and mic switchboards existed before Stokowski's involvement, and more than one mic was used in film sound tracks, the levels and tonal balance were predominately controlled from a technical point of view. It was Stokowski that brought music into the control room and married music and engineering to mark the beginning of mixing as a musical/technical art form.

Bibliography

Cameron, James, R. *Motion Picture—With Sound.* New York: Cameron Publishing Corp., 1929.

Cass, John L. "The illusion of Sound and Picture." *Journal of SMPTE,* February, 1930.

"Employ Novel Methods to Put Orchestra on Air," *Philadelphia Inquirer,* October 6, 1929.

Horn, C.W. Radio Address. Chicago, January 3, 1923.

McGinn, Robert E. "Stokowski and the Bell Laboratories, Collaboration in the Development of High Fidelity Sound Reproduction." Paper presented at the annual meeting of the History of Science Society, New York, December 29, 1979.

Smith, Stephen G. Jacket notes for "Leopold Stokowski: The Early Years." Recording LS3.

"Sound Waves 'Rock' Carnegie Hall as 'Enhanced Music' is Played." *New York Times,* April 10, 1940.

Stokowski, Leopold. "New Horizons in Music." *JASA,* July, 1932.

Wallace, Bob. National Public Radio interview with Stokowski. 1965.

Whyte, Bert. "My Friend the Maestro." *Audio,* January, 1976.

A CLOSER LOOK

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tronically balanced inputs and outputs. The POD input and output connections are via screw terminals (which makes sense for a permanent installation), whereas the Model 2800 itself has both balanced XLR and unbalanced phone jack connections. Nominal operating level is about 0 dBm, and rated maximum output +20 dBm, although Rimkeit said they should be capable of +24 dBm out. The unit works faster than I can...about 30 seconds for automatic equalization to your specified *ideal* curve. If you've got a room to tune, and no time to waste, this item deserves *your closer look.*

Gary Davis owns Gary Davis and Associates, a firm which has produced instruction manuals, catalogs, advertisements, and newsletters for the audio field over the last 12 years. Davis has worked as an electronics technician, is a member of the Audio Engineering Society and author of the CAMEO dictionary.

DATAFILE *info. sources/new literature*

New Catalog of Tools and Test Equipment from Jensen

A new catalog of tools and test equipment is offered free by Jensen Tools Inc. Illustrated in full color, the 160-page catalog contains more than 1,000 items of interest to engineers and technicians in the telecom and communications industry.

Two new sections feature supplies and equipment in support of fiber optics and wire/cable systems. Other major categories cover metric tools, power tools, lighting and optical aids, wire strippers and crimpers, tweezers, pliers, screwdrivers, soldering/desoldering equipment and supplies, shipping containers, test equipment, static control products, computer accessories, and a complete selection of dedicated tool kits and cases.

The catalog also describes Jensen's custom kit building service. Kits may be designed to any specification and shipped to all parts of the world.

Circle 8 on Reader Response Card

New Brochure from FSR, Inc. Features Accessories

FSR, Inc. has released a new accessory brochure that describes products that can add the finishing touches to any installation. Among the products that are described are interconnect cards that feature screw termination, proper termination for audio lines, tranformer cards for mounting Altec 15335(15 to 15 K) or 15905A(600 to 15 K) transformers or Op Amp 10 kHz tranformer.

For audio wiring the company has the J-3 Audio Combine and the J-5 Audio Junction. Rack Demarcation cards are also available in model numbers B-1A or B-4, B-5. In addition the B-3, a power supply distribution card, is available.

The sheet also describes floorboxes, gooseneck microphones, power distribution card, cue and monitor speakers, VU-meter interface cards, image reversing mirrors, power supplies, and a remote audio telephone signal.

Circle 9 on Reader Response Card

1986-1987 *Locator* Now Available From the ERA

The Electronics Representatives Association (ERA) has announced publication of the 1986-87 edition of its annual *Locator* of electronic industry manufacturers' representatives. The *Locator* is designed as a service and reference source for electronic industry manufacturing sales personnel and includes listings of professional representation available throughout the United States and in Canada, Europe, the Caribbean, and the Far East.

The *Locator* contains listings of ERA's 2,200 representative member firms by geographic territory. Pertinent data for each firm includes: territory covered; types of products the firm handles; number of personnel; branch offices; and additional facilities and services. Officers and key personnel of each representative firm are also listed as contacts.

The *Locator* will also be available free throughout 1986-87 at the many electronics industry trade events at which ERA exhibits. Copies may also be requested by mail or telephone.

Circle 10 on Reader Response Card

Electronic Specialists, Inc.'s VCR and Video Protection Catalog

Electronic Specialists Inc. has released its 40-page color catalog which describes products for VCR and video equipment protection. Uninterruptible power supplies, line conditioners, spike suppressor/filter combinations, equipment isolators, and AC power interrupters are listed.

Included are tutorial sections describing various video problem situations and corrective action to be taken. The catalog describes numerous standard, off-the-shelf products, custom designs and Hot Line problem solving. Catalog 861 is free

Circle 11 on Reader Response Card

SUPPLIES•SERVICES•SUPPORT

Berk-Tek's Woven Flat Fiber Optic Cable

Berk-Tek has introduced fiber optic cables that are said to out-perform conventional copper cables when space savings, weight, and immunity to EMI/RFI are critical.

When woven into a flat construction, the cables offer a more flexible and extremely low profile harness. All sizes of fiber and various outer jacketing materials are available along with a wide variety of connectors for terminated assemblies.

Circle 1 on Reader Response Card

ESE Introduces Audio Level Indicator

ESE has introduced a new audio level indicator, the ES 216, designed to combine the most desirable features of the mechanical type audio meter with the most desirable features of the LED type audio meter.

Featuring a semi-circular array of green, yellow, and red LEDs, the ES 216 fits into the same size cutout as typical mechanical audio meters. The ES 216 is said to be more accurate and easier to read than mechanical type audio meters, but it does not require the user to adjust visually to the less common linear configuration used in other LED type audio meters.

Circle 2 on Reader Response Card

Pirelli Introduces Compact Undercarpet Cable System

Pirelli Cable Corporation has introduced a non-proprietary undercarpet cable system for power, data, and telephone connection.

The system, which is U.L. listed, can be used with equipment produced by a variety of manufacturers. It requires few floor fittings and its universal transition block fits most flush undercarpet cable wall boxes. An adapter block is concealed by a decorative cover, and toolless connectors are supplied for voice and data communications.

The undercarpet cable system, including cable for normal power, computer power, telephone and data connections, is compact and entails low-cost installation. Its three, four, and five conductor floor fittings offer three power phases, two isolated-ground options, and two split phase options.

Circle 3 on Reader Response Card

David Clark VOX Belt Station For Hands-Free Communications

David Clark Co. has made available a voice operated transmission (VOX) Belt Station which permits hands-free operation of the company's communication systems.

Called the V3412, the modular unit features conveniently located controls to adjust volume and VOX sensitivity, manual push-to-talk override button, RFI shielded cable, and belt clip.

The unit can be added to existing systems without modification of the system and works with all high impedance headsets produced by the David Clark Company.

Circle 4 on Reader Response Card

Eliminate Spikes and Surges With The Glitch Grabber

GC Electronics introduced the Glitch Grabber AC Line Filter and Surge Protector with six-foot power cord.

The Glitch Grabber is an important accessory to electronic equipment: computer, stereo system, television, VCR and photographic equipment. With a peak surge rating of 19,500 amps, your equipment is safe from frequently occurring spikes and surges with the Glitch Grabber. The Glitch Grabber allows clean 120 VAC energy to pass through, eliminating the damage to sensitive equipment that can be caused by EMI/RFI noise. A neon lamp indicates properly grounded outlet. The Glitch Grabber is constructed with all U.L. recognized components and backed by five year warranty. A wall mount model is also available. Suggested resale is $49.95.

Circle 5 on Reader Response Card

Upgrade Phone Systems With Audiocom's Promotions on Hold

Audiocom, Inc. has introduced a telephone message on-hold service, "Promotions On Hold," which can be sold as an effective add-on to business telephone systems.

The Promotions On Hold service plays company messages interdispersed with entertaining music for all calls placed on hold.

The service is designed to improve customer service, introduce new products and services, cross-market existing products and services, decrease caller hang-ups, and increase revenues. Audiocom provides all services necessary to produce the messages. The only prerequisite of the Promotions On Hold system is that the telephone system has music-on-hold capability.

The company also distributes an Automatic Call Sequencer for businesses with a high volume of incoming calls. The sequencer automatically answers calls and places them on hold in sequence.

Circle 6 on Reader Response Card

Midland LMR Introduces Auto Phone Patch For Repeaters

Midland LMR's Model 70-2070 repeater phone patch provides automatic interconnection to the telephone system from mobile units equipped for DTMF signalling. Mobile operation is half-duplex, and the mobile push-to-talk activation controls the audio paths through the interconnect.

The new phone patch will disconnect after three minutes (preceded by a warning tone), and automatically disconnects if no mobile activity occurs within one minute. On-board LEDs indicate *key* or *connect* conditions as well as *star* or *pound* decoding for tests.

Circle 7 on Reader Response Card

CALENDAR OF EVENTS

DATE BOOK

DATE	EVENT/COMMENT	LOCATION	CONTACT
November 13-16	**81st Annual AES Convention** Papers and exhibition.	Los Angeles, CA	AES (212) 661-2355
November 18-19	**Sound Engineering Seminars.**	Orlando, FL	Syn-Aud-Con (714) 728-0245
November 19-20	**Fiber Optics in Plain English.**	San Francisco, CA	Clifford Inc. (802) 234-9921
December 4-5	**ESSC Regional Conferences** Products and seminars.	Secaucus, NJ	Bob Barba (312) 593-8360
December 8-9	**ESSC Regional Conferences** Products and seminars.	New Carrolton, NJ	Bob Barba (312) 593-8360
December 10-11	**Fiber Optics In Plain English.**	Dallas, TX	Clifford Inc. (802) 234-9921
January 7-9	**Fiber Optics Workshop.**	Lake Buena Vista, FL	University of Central Florida (305) 275-2123
January 21-23	**Solid State Electronics for Non-Electrical Engineers** Course presented by The Center for Professional Advancement.	E. Brunswick, NJ	The Center (201) 238-1600
February 9-11	**Solid State Electronics for Non-Electrical Engineers** Course presented by The Center for Professional Advancement.	Houston, TX	The Center (201) 238-1600
February 26-28	**Commtex International '87.**	Atlanta, GA	Bobbie Hunt (703) 273-7200
March 16-18	**Technical Report Writing** Course presented by The Center for Professional Advancement.	E. Brunswick, NJ	The Center (201) 238-1600
March 30-April 1	**Technical Report Writing** Course presented by The Center for Professional Advancement.	Boca Raton, FL	The Center (201) 238-1600

FACES AND PLACES

Renkus-Heinz Names Weisman as National Sales Manager

Renkus-Heinz has announced the appointment of Irv Weisman to the position of national sales manager, Smart Systems Division. He will be responsible for coordinating the Smart Systems dealer network nationwide. Weisman has been involved in the professional audio industry since 1973, when he introduced BGW Systems to the industry. He was involved in the market development of Crest Audio in the late 1970s and the recent introduction of B.E.S. to the commercial sound industry.

Stentofon Appoints Caldwell as Business Manager

Stentofon Communications, Inc. has announced the appointment of Douglas Caldwell to the newly created position of business manager. Caldwell's responsibilities include sales analysis and projections plus the development of a company-wide information system/data base for enhancement of customer service activities.

Caldwell comes to Stentofon with a master's in Business Administration and experience as a project analyst with an investors group and as general manager of a continuous-forms manufacturing plant.

DOUGLAS CALDWELL

ALLEN W. DAWSON

Siecor CEO Dawson Elected USTSA Treasurer

Allen W. Dawson, chairman and chief executive officer of Siecor Corporation, was recently elected treasurer and a member of the Executive Committee of the United States Telecommunications Suppliers Association (USTSA).

First named to the USTSA board in 1985, Dawson will serve a one-year term as treasurer of the Association, which represents over 600 manufacturers and suppliers of telecommunications equipment.

Stolte Named Product Design Engineer at Altec Lansing

Brandon Stolte has been named product design engineer at Altec Lansing Corp. He will be involved in product design, control panel layouts, and packaging design.

Stolte has a bachelor's degree in Industrial Design from the University of Illinois at Urbana-Champaign. Before joining Altec Lansing, he was with Peavey Electronics Corporation. He is a member of the Consumer Products Technical Group of the Human Factors Society and an associate member of IDSA (Industrial Designers Society of America).

dbx/ADC Appoints Menozzi Vice President, Marketing & Sales

David G. Kennedy, president of dbx/ADC has announced the appointment of Alfred J. (A.J.) Menozzi as vice president, marketing and sales.

Menozzi has held sales executive positions for over a decade with major consumer electronics firms. From October 1984 until he joined dbx/ADC, Menozzi was general manager, northeast region for Toshiba America, Inc. He was responsible for all of Toshiba's sales, marketing and administration in the northeast quarter of the U.S.

Jerry Crane Joins V Band as VP, Product Design and Development

V Band Systems, Inc. has announced the appointment of Jerry D. Crane as vice president, product design and development.

Prior to the V Band appointment, Crane was employed by Data General Corporation where he held the positions of product manager, workstation development, and senior manager of terminal development.

Crane holds a BSEE from the University of Texas. He has been elected to Tau Beta Pi, Eta Kappa Nu and Phi Kappa Phi honor societies.

SOUND & SECURITY

Sabel Completes Fame City Installation

Sabel Commercial Electronics recently completed the installation of a sound reinforcement system and a CCTV surviellence system in "Fame City" an indoor family entertainment center in Houston, TX.

Completed in June, 1986, the self-contained Fame City Complex consists of a miniature golf course; a 40 lane bowling facility; three movie theatres; a teen disco; a children's center and game area; numerous restaurants and shops; plus a water park adjacent to the complex.

According to James Hardcastle, sales manager of Sabel Commercial Electronics, "We worked closely with the owners for about four months before starting work on the interior of the building. We gave them a one-year guarantee on the whole system, so we chose equipment that was reliable."

Among the equipment installed was Toa's 900 series amplifiers, PM-600U paging microphones, BA-400 cassette players and Toa's HS-215 two-way speakers were installed in the Main Street indoor courtyard. In the bowling area Soundsphere 110 loudspeakers were installed because "they gave better dispersion and sound coverage for the bowling alley that has a high noise level," said Hardcastle. At the golf area, Lowell 2215 loudspeakers were installed. In the laird, there is a 26-inch color Panasonic monitor that has a character generator for ads and other messages. The system also has the capability to send video from any camera mounted in the park or from a satellite and beam it inside the park. The bowling area has an AMF color monitor. The cameras can show not only the player's score but audio and video output from other sporting events. Sabel also installed an RCA computer-controlled video system the TC1600 and Winsted racking equipment.

Altec Awards Top Dealers

Altec Lansing Corporation held its annual Concertmasters business conference in Puerto Vallarta, Mexico on October 23-27. Each year the corporation awards its industrial/professional contractors and distributors who have attained a predetermined performance and sales goals with a vacation in an exotic locale. This year over 50 of Altec's top contractors, distributors and guests came from all over the world. Also in attendance were the Altec Lansing corporate executives and full sales staff.

According to Gary Rilling, vice president of marketing, strategy and Altec's position in the marketplace were two of the areas that were discussed at the four day conference.

PIERCE CLOSES CA OFFICE

Peirce-Phelps, Inc. has closed its Costa Mesa, CA, sales office, citing a decline in West Coast systems business.

"We view the decline as part of a general cutback in capital spending due to uncertainties regarding the tax revisions and the state of the economy," Henry S. Grove, vice president of the Video Systems Division of Peirce-Phelps, said. "We expect a turnaround, but there is simply no way to predict when."

Phil Gantt, who moved to the West Coast to head up the office, has elected to remain in California and seek other opportunities. Jeff Wilson will continue to provide maintenance services for Peirce-Phelps' teleconferencing systems in the West.

Sales will be taken care of by the company's Philadelphia headquarters.

***PLEASE SEND** all your contracting company's news to:*
Contracting Close-up
Sound & Communications
220 Westbury Ave.
Carle Place, NY 11514

THEORY & APPLICATION

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can protect against before it fails.

Furthermore, when utilized with audio and video equipment, the spike protectors alone can introduce additional noise into the line. It is advisable that a noise filter be provided following such spike protection. Noise filters, generally LC multipole low-pass filters, are usually, but not always, included in better surge protectors.

Each circuit for audio or video equipment should include an independent grounding means. The ground pin on the outlet must not be connected to its conduit but instead be connected to an insulated 12-gauge copper wire carried through to the earth ground point for the system, which should be a copper pipe driven at least six feet into conductive soil, in conformity to local electrical codes.

When connecting equipment to such an outlet, one must remember that circulating currents can be set up as shown in *Figure 1.* One finds apparent conflict between safety requirements for grounding and requirements for maintaining low noise.

Sometimes this can only be corrected with the use of isolation transformers, however, in doing so it is difficult to conform to electrical codes. Any nonconformity in connection of safety grounds must be regarded as extremely dangerous. It is better to carefully plan your signal grounds so that low-level points do not ride on power grounds. Electrical plans should also be examined beforehand to see that power grounds for the sound equipment circuits are not physically separated in their runs to the service panel. Conduit runs around opposite sides of the building make excellent pick-up coils for every disturbance in town.

Good power system design does not require a sophisticated knowledge. It only requires a realization that power circuits are inherently unstable and grounds are never at ground potential—and an ability to convince others accordingly.

Daniel Queen has been the head of Daniel Queen and Associates, an acoustical consulting firm, since its inception 20 years ago. He is also the chairman of the AES Technical Council as well as a member of *Sound & Communications'* Technical Council.

LAB TEST

(continued from page 37)

down. Because there are no mechanical stops in the remote, it is break-proof and can even be mounted in a publicly accessible location. For simplifying installations, the remotes may be daisy-chained. The preset and priority levels may be set by a single-pole switch or logic ground closure without any remote at all. Neither noise nor distortion could be detected throughout the tests when we rapidly changed levels or plugged in or disconnected remotes from the master.

It is clear that Oxmoor paid careful attention to detail in the design and the construction of both the DCA-2 and its RC-16 remote controls. From a service viewpoint, the boards are elegant, both well-spaced and properly labeled for easy repair if ever necessary.

Though all the applications are too numerous to list here, it should be obvious that these devices will find their way into media and boardroom systems where precise pre-set levels are desired with key-lock security and added luxury.

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We view the contractor market as a serious growth opportunity for Crest Audio, and with this in mind, we are creating a separate division for which we need an experienced individual to manage all phases of the Marketing and Sales.

If you have extensive knowledge of the contractor market, good understanding of amplifiers and systems, and have a successful sales and marketing background, Crest Audio can offer you a career opportunity with excellent potential. We're seeking a hard-working individual who has the ability and dedication to build Crest Audio into as strong a position in the contractor market as we now enjoy in the sound company market.

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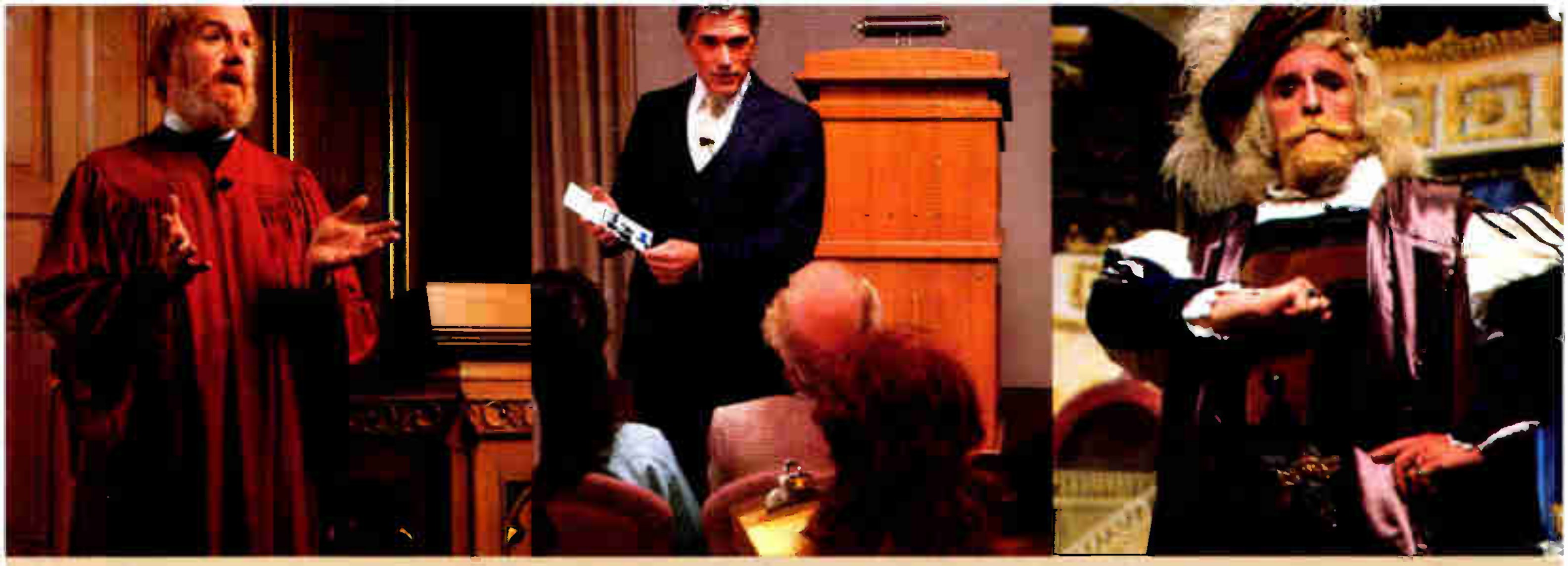

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