SYNERGETIC
SYN AUD CON
AUDIO CONCEPTS

newsletter

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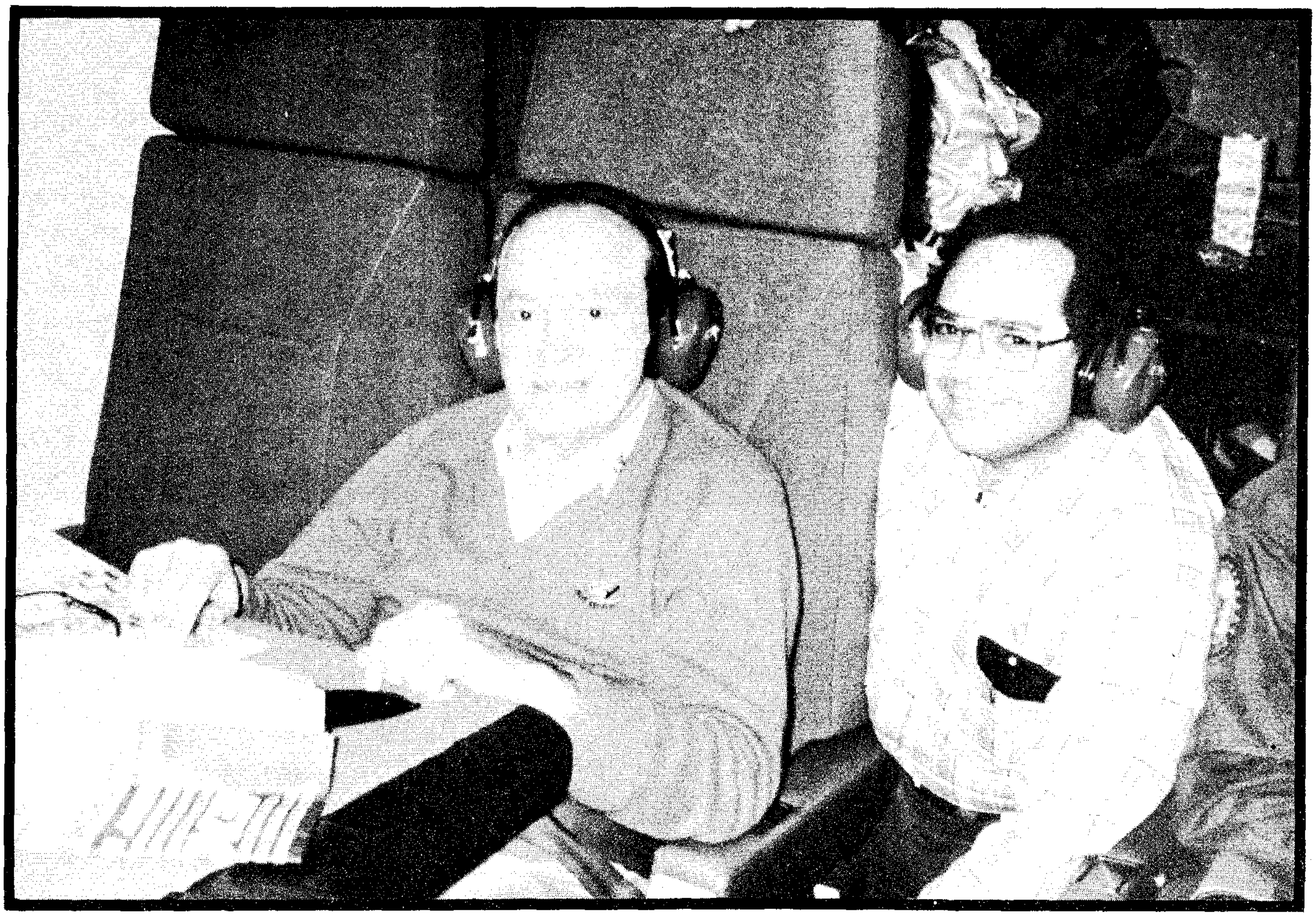
Volume 11, No. 4
Summer 1984

SYNERGETIC
Working together; co-operating, co-operative

SYNERGISM
Co-operative action of discrete agencies such that the total effect is greater than the sum of the two effects taken independently.

EXCHANGE OF IDEAS

I met a man with a dollar We exchanged dollars I still had a dollar	I met a man with an idea We exchanged ideas Now we each had two ideas



LARRY ESTRIN AND FRIEND
(Wearing Syn-Aud-Con Hearing Protectors)

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THE BIG GUNS

Larry Estrin and Tom Durrell, Los Angeles, CA., have a bigger rifle than Syn-Aud-Con. Not only that, they travel with a better storyteller than Don.

The ratio of: $10 \text{ Log} \left(\frac{16"}{0.460"}\right)$ = 15.41 dB impresses any shooter.

Note, however, that Syn-Aud-Con's colors were present and we'll give a set of free hearing protectors to anyone who can show a higher caliber rifle or entertainer.

The front cover photo resulted from Bob Hope's trip to Lebanon to entertain the troops there last Christmas. Larry and Tom handle these kind of audio challenges with practical ease. ◆

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THE SUNN ADS 4000 SERIES DELAY

Larry J. Lynn, president of Sunn Musical Equipment Company of Tualatin, Oregon, has informed us that the following models of their remarkable ADS 4000 series will be in full production in September of this year. These are the even further enhanced versions of the unit we have been demonstrating in our recent classes and workshops for the signal alignment of loudspeaker devices covering the same frequency range but from differing acoustic centers.

ADS 4002 - Two Output Model (reference and one delay)	LIST:	$2,599.00
ADS 4003 - Three Outputs (reference and two delay)		$2,899.00
ADS 4004 - Four Outputs (reference and three delay)		$3,199.00
ADS 4005 - Five Outputs (reference and four delay)		$3,499.00
ADS 4006 - Six Outputs (reference and five delay)		$3,799.00

Available factory assembled in any of the above configurations and delay output modules can be ordered individually to expand unit at later date.

SDM 4001 - Signal Delay Output Module $ 300.00

Four preproduction units will be available in July and production units available in September.

Remembering the severe penalties in both tonal perception and loudspeaker *coverage* deterioration when signals do not arrive simultaneously, the Sunn ADS 4000 series is a real bargain as well as an ideal technical solution.

Using Charles Bilello's improved resolution ETC approach, the TEF® Analyzer provides 15.7 usec resolution which is sufficiently close to the 10 usec per step of the Sunn ADS as to allow direct measurement to within one step of the correct value. We have demonstrated in our classes that 30 usec is a clearly audible signal delay and that 20 usec can be detected by trained listeners. It is our present belief that 10 usec steps is an ideal increment for a signal delay correction device.

We are looking forward to reports from early users of Sunn's device as to application chosen and results obtained. We feel this unit bears a similar relationship to TEF that equalizers had to 1/3 octave real time analysis. ♦

SAN FRANCISCO - MARCH 14-15, 1984

SYN-AUD-CON WORKSHOPS 1984-85

LEDE™ WORKSHOP: September 11-13, 1984 - Nashville, Tennessee

Just one glance at the tentative outline Neil Muncy, LEDE Workshop Chairman, has prepared for the LEDE Workshop in Nashville, Tennessee, on September 11-13, 1984, reveals an out-of-the ordinary event.

PRELIMINARY OUTLINE OF TOPICS TO BE COVERED IN THE L.E.D.E. WORKSHOP

DISCUSSION

Why L.E.D.E.? Comparison of best features of concert halls with past H.F.D.R. control room designs. Why NOT volcanic rock, basstraps and compression ceilings? The importance of realizing a useful I.T.D. Getting enough diffusion. Psychoacoustics.

LISTENING SESSION

Participants listen to their own recordings in the Acorn control room.

WHAT IS L.E.D.E.?

Develop an updated definition of L.E.D.E.

HOW TO IDENTIFY AN L.E.D.E. ROOM

What should you expect to SEE?
What should you expect to HEAR?
What should you expect to MEASURE?

DEMONSTRATIONS

The effect of a strong early reflection (covering one front sidewall with a hard material).
Polar patterns of monitor loudspeakers.
Removing the diffusion (covering the RPG's with plywood).

MEASUREMENTS

I.T.D., decay time, on & off axis monitor loudspeaker response, using both TEF and conventional techniques.

HANDS-ON DESIGN PRACTICE

Basic room shape--starting room ratios--bilateral symmetry--floor plan for functional layout--traffic flow--equipment lists--provision for future expansion--lighting--HVAC--isolation from outside noise--aesthetics.
Reflection control--ray tracing floor plan & elevation drawings to identify first order reflections.
How Dead is Dead? Construction of the front shell--selecting the proper absorbent material.
How Live is Live? Construction of the rear of the room--obtaining enough diffusion and low frequency absorption.

FIELD TRIP TO MASTERMIX & SKYELABS BUS

Measurements and listening in two (possibly three) different L.E.D.E. environments.

Past workshops have concentrated on the theory underlying LEDE and the pioneering approaches used by early practitioners. This workshop is intended to begin the formalization of the best design practices established by actual performance in the field.

The workshop will be held in the first control room in the world to use our original suggestion for diffusion. The unique Quadratic Residue Diffusor (QRD) by Dr. Peter D'Antonio is based upon Manfred Schroeder's work on the subject. Dr. D'Antonio is also a staff member for this workshop and will have on hand special "Gobos" consisting of a diffusive side, reflective sides and an absorptive side for use in studio environments.

Robert Todrank, the designer of Acorn's LEDE control room is the host for the workshop. The staff includes Neil Muncy - Chairman, Dr. D'Antonio and Don Davis. Special assistants include the pioneer practitioners of this technique: Chips Davis, Russ Berger and Glenn Meeks. The ETC for the Acorn control room is ideal. The density and temporal distribution of the decaying sound field, thanks to the special diffusors, is that normally encountered only in large concert halls. The decay rate is, of course, faster than a large hall but under exceptional control. We believe that both the sound quality and the objective measurements represent a new standard for control rooms.

TEF® Application & Instrumentation Workshop: Nov. 13-15, 1984 - Atlanta, GA

Originally we scheduled the workshop for September 1984 in Elkhart, Indiana. We recently heard that Don Keele had gone to work at Tecron to develop new programs for the TEF analyzer and we knew we had to wait for his output to share in the workshop.

The Workshop Chairman is Don Eger. The emphasis will be on application and interpretation of measurements. Dr. Patronis and Don Davis will contribute to this aspect of the workshop. Don Keele will share his work at Tecron with the members of the workshop.

Concert Hall Workshop (tentative): December 1984

If we can work out the details, we will hold the workshop in mid-December. We have determined that it is not feasible to duplicate here the unique experience that Mr. Peützt provided us in the Netherlands in 1983 -- the same orchestra, the same conductor, the same repertoire in three different concert halls in three days.

Continued next page.....

However, we did learn from our one-day "on location" measurements at Boettcher Concert Hall in Denver that three different concerts in three different concert halls would provide a valuable learning experience *IF* we can attend a dress rehearsal in the morning, measure in the afternoon and attend the concert in the evening. Where can this be accomplished? What area of the country would provide three concert halls (excellent, good and maybe not so good acoustics), all within easy travel?

How many excellent concert halls are there in the United States? Boston Symphony, for sure. But will the unions in the East allow use of concert halls in Boston and New York without fearsome expense?

Mr.Peütz has agreed to conduct the workshop if we can work out the details. If you can help--let us hear from you.

Business Organization & Management (tentative): Feb. 1985 - So. California

Workshop Chairman: Victor M. Hall of Communications Company, Inc., San Diego, California.

Grounding for Audio Systems: Spring 1985 - Minneapolis, Minnesota

Workshop Staff: Edward G. Lethert, Jr., of Michaud, Cooley, Hallberg, Erickson & Associates, Minneapolis, Minnesota, and Christopher R. "Topper" Sowden of Joiner-Pelton-Rose, Inc., Dallas, Texas.

Installing & Troubleshooting the Sound System: Late Spring 1985 - Eastern USA

Workshop Chairman: Philip B. Clark of Diversified Concepts, Inc., Marcellus, New York. ♦

SYN-AUD-CON SEMINARS AT RESORTS

Honolulu, Hawaii - December 3-4, 1984

Our very special "grads" from Hawaii have encouraged us for years to hold a class there. They tell us that they will provide all the necessary test and audio equipment for the class. We will bring our books and manuals plus a few special items.

Rodney Kobayakawa, who has attended two TEF Workshops plus a regular 2-day class, is encouraging us to hold a one-day basic-basic class before the two-day class. We're giving it serious consideration.

The Institute for Noise Control Engineers is holding its convention the same week, immediately after our class. Victor Peütz is active in INCE and will be attending. Mr. Peütz has had his TEF analyzer since March 1984. We are eager to see his measurements and to hear of the insights he has gained from his work with the TEF analyzer.

Dana Point, California - February 6-7, 1985

Many Syn-Aud-Con graduates have expressed a desire for an occasional class at the Marina Inn in Dana Point, California. They remember the relaxed atmosphere, the excellent restaurants, and the myriad shops and recreational facilities that allowed families traveling with them to enjoy a unique Southern California seaside resort area while they attended a Syn-Aud-Con class.

We have, therefore, scheduled a class February 6-7, 1985, at the Marina Inn for those of you wishing to experience the advantages of this location. This class will be the latest version of our two-day Syn-Aud-Con class. Because attendance at this class must, of necessity, be limited due to the size of the meeting room at the Marina Inn, we suggest that you register early to be ensured of securing a place in this particular class. ♦

"BIG JOHN"

"Big John" Laberdie is a particularly competent audio/video computer expert who has not been previously known for sartorial elegance. The professional looking distinguished gentleman in the tuxedo is Big John. We are told he still had his sandals on. In any case, this evidence qualifies him for listing on this year's best dressed list of Dave Andrews' Associates. ♦

PRESBYCUSIS, SOCIOCUSIS AND NOSOCUSIS

Hearing loss (HL) discussions use the terms:

Presbycusis
Hearing loss attributed to the aging process.

Sociocusis
Non-work noise induced hearing loss.

It is our *opinion* that most hearing losses occur recreationally.

Nosocusis
Non-noise related otological disorders, such as, middle ear disease, ears impacted with wax, otological trauma from blows to the head, use of ototoxic drugs, or sensorineural disorders.

It is no wonder that a zero percent articulation loss for consonants is an impractical goal.

A fascinating article by Karl D. Kryter in the *Journal of the Acoustical Society of America,* June 1983, pp. 1897-1917 covers these subjects in some detail in easy to understand English.♦

WOMEN IN SYN-AUD-CON CLASSES - FALL/WINTER 1983

NEW ADVENTURES

Bill Watkins - Over two decades at the Fraizer Company, Bill is now at work at J. W. Davis & Company.

Jay Mitchell - A Dr. Patronis graduate student from Georgia Tech, Jay is working at J. W. Davis & Company for the summer. He is keeping the TEF® analyzer hot.

Don Keele - Formerly of Electro-Voice and JBL, Don has joined the Tecron Division of Crown International--not to design horns, but as a programmer par excellence for the TEF analyzer.

Bill Raventos - Many years protege of Lou Burroughs at Electro-Voice, Bill is now at Crown International working with PZMs.

Geoff Langdon - Of AKG and more recently of Sennheiser, Geoff has joined Rupert Neve, Inc. ♦

"USED" CLUSTER COMPUTERS FOR SALE

We had twenty Community Cluster Computer packages for our Loudspeaker Array Design Workshop in Nashville. Members of the workshop used the Sphere packages to design the cluster for West End United Methodist Church in Nashville.

Members of the class purchased all but eight of the packages. We checked them out carefully and added two water soluble pens as a bonus.

Normally the Cluster Computer sphere program sells for $195 plus $4.00 handling. If you would like one of our "used" complete programs, we will ship prepaid to you for $180. Make your check to Community Light & Sound but order from Syn-Aud-Con. ♦

LIGHTNING PROTECTION INSTITUTE

Steve Bushelman, Jr., in our Orlando class in 1983 said he had recently taken a rigorous course and exam on lightning protection. (Florida is known as the "Lightning Capital of the United States.") For this reason, EPCOT did careful research before building EPCOT. EPCOT Center uses UFER grounds. Ground bus in the electric equipment room of every building is tied to the UFER ground system and provides a common bonding point for system grounds, equipment grounds, water pipe grounds and the lightning protection systems grounds.

While we were in Orlando, Steve gave us a piece of literature from the Lightning Protection Institute and we wrote for more. We think it is useful and that you should know more about the Lightning Protection Institute. We are reproducing here the LPI Code of Ethics.

The literature we received included a statement by Edward A. Lobnitz, P.E., of Tilden, Lobnitz & Cooper, Inc., who was then General Chairman, Professional Division, Lightning Protection Institute.

> *The construction professional, particularly the design engineer, needs to become knowledgeable and involved in lightning risk evaluation, in lightning protection system design and layout, and above all, in lightning protection system quality control.*

Write LPI if you want more information on lightning protection. ♦

LPI Code of Ethics

As a Member of the Lightning Protection Institute, I agree to . . .

1. Promote development and use of standards assuring positive lightning protection.
2. Promote good will between the public and this association's members.
3. Employ skilled and ethical lightning protection installation personnel.
4. Use, specify and promote lightning protection components and systems meeting prevailing code requirements.
5. Promote the use of just and explicit contracts under the best guides for consumer protection.
6. Promote professionalism and work to correct any abuses by members of this industry.
7. Strive to advance the Lightning Protection Institute, its standards, its programs, and its memberships.

LIGHTNING PROTECTION INSTITUTE

48 N. Ayer Street Harvard, Illinois 60033 815/943-7211

FOR SALE - FFT

Syn-Aud-Con's Gen Rad 2512 FFT Analyzer is for sale. This is the unit you have seen us use in the classes in past years. Recently recalibrated by Gen Rad and in perfect working order. New units of this model currently sell for $18,000.00 plus. Your price: $5,000.00 - complete with detailed operating manual. ♦

SHOPPING FOR SIGNAL PROCESSORS

by Gerald Stanley

Technology which is new to the masses is an abundant opportunity for fraud and deception. Digital signal processing is proving no different on this point than did the introduction of electrical instruments into the medical practices. When the public heard that electricity had analytic and therapeutic values, panels of meters and dials were in abundance in every quack's office.

A sense of magic is the quack's most potent medicine. Such abuses are unfortunate in an excessively cynical world. Digital signal processing is not magic but is very useful when properly applied to the application at hand.

A brief tutorial on some of the major points is in order.

1. The term FFT is short for Fast Fourier Transform which is the name of an algorithm for computing the Discrete Fourier Transform or DFT. The DFT is in turn a time restricted sampled version of the continuous Fourier transform. The Fourier transform is a mathematical relationship between the alternative views of a signal as a collection of frequencies or as a time waveform. Both representations contain the same total amount of information, they are just alternative ways of saying the same thing. The Fourier transform is the road map between the two representations. The DFT uses a finite collection of frequencies or time samples to represent the signal. The FFT is simply a faster way to organize the computation of the DFT and thus is called a Fast Fourier Transform. It's not magic any more than long division is magic compared to successive subtraction to perform division.

2. The TDS transform is a higher dimensional form of the Fourier transform. In fact the Fourier transform is a special degenerate case of the TDS transform. They are not in fact identical in form fit and function. FFTs are performed within the TEF world to produce displays such as the Energy Time Curve or ETC.

3. In digital signal processing systems, a very important parameter is the number of bits of data gathered per data sample. Each added bit increases the number of states that can be represented by two fold or 6 dB. In other words, a system having 12 bits of data conversion is not 50% more accurate than an 8 bit system, but is rather 16 times as accurate. The TEF-10 is a 12 bit system for both signal input and output.

   A 12 bit system has a dynamic range limit of 75 dB based on the quantizing effects of 12 bits. This compares favorably with studio tape recorders of the analog era and can thus be readily understood as being compatible with the dominant range of hearing. This can definitely not be said for an 8 bit system.

4. When testing physical systems using active tests, i.e., tests that supply their own stimulus to observe a response, the signal to noise is directly related to the amount of stimulus energy. The more energy used as input to the system, the more will be available for analysis in the output of the system.

   The use of impulses for testing systems is one of the poorest signals known for energy content despite its conceptual simplicity. To impulse test using a single channel FFT and test a bandwidth of 20KHz requires a pulse of approximate width of (1/20,000) seconds or 50 microseconds or less.

   By comparison with a one second 20KHz TDS sweep, the TDS sweep has approximately 20,000 times the energy of the impulse. To acquire the test S/N or energetics of the TDS will then require the impulse test to be repeated with signal averaging no less than 20,000 times. If the FFT is extremely slow such as 18 seconds, this will require over four days of measuring time to get what the TEF-10 will gather in one second. To spend over four days to get an 8 bit result would be a crime of absurd proportions.

5. An all important principle of joint-domain representations (time vs frequency response), is that the resolution can at best be such that the product of the time resolution and the frequency resolution will be unity. That is to say that the knowledge of time resolution to one microsecond accuracy will demand that the frequency not be known to better than 1MHz resolution. Likewise a frequency resolution of 20Hz will limit time resolution to 50 milliseconds. This trade-off may be readily performed with TDS techniques by the choice of instrument bandwidth. The time resolution is B/S where S is the sweep rate in Hz/sec and B is the bandwidth in Hz. Therefore the frequency resolution is S/B.

   When making conventional FFT joint-domain measurements of either the single channel or two-channel type, the time resolution is set by the length of the data record gathered, or its effective time window. A 1024 point FFT to display a 20KHz bandwidth must gather a 20 millisecond data record. Unless a very narrow (in time) window function is applied to the data record, this is the de facto time resolution of all joint-domain displays produced by such instruments regardless of the cost. The use of a very narrow window (not found on any of the commercial units) discards input data and greatly increases the noise of the FFT as quantizing noise becomes severe. Whether single or dual channel, the prospects for good S/N and time resolution without using TDS is theoretically bleak. This is the reason that all of the publicly displayed joint-domain displays produced by FFT methods have not shown sufficient time resolution to do good speaker alignment. Little wonder that the bat uses TDS methods to capture flying insects in the dark of night rather than use commercial FFT analyzers. Nature is always stealing the best of our inventions.

Continued on next page...

6. To maximize the accuracy of an analysis of system response the foreknowledge of the test signal must be used when observing its response in the output of the system under observation. In other words, if a farmer planted wheat in a field, it would be foolish at harvest time to be harvesting with a potato digger. He would have no need to plow through all of the dirt to find the wheat which grows at the top of the wheat stalk.

   This is the fundamental concept of the matched filter. TDS techniques use a filter which is matched to the signal that was input to the system to observe the output. The analyzer is looking for a sinusoidal sweep signal which is precisely delayed from the test signal. The FFT analyzers on the other hand are digging through every bit of dirt in the test environment.

7. Energy Time Curves (ETCs) are plots of energy shown on a log energy (dB) vertical scale with linear time on the horizontal axis. The dynamic range is potentially very large when using a dB scale. The information found in the ETC is the same information as found in the classical impulse response if the impulse response magnitude is plotted with a dB scale. This is not the traditional display for the impulse response as it is typically shown as only its real part with a linear vertical amplitude scale and linear horizontal time scale. It is not possible to see the range of features in the classical impulse response display that is needed to analyze the character of reverberant sound fields.

   Often the promoters of FFT equipment will attempt to convince potential users that the linear display is just as good without mentioning that the noise floor of their measurement is too high to see much of the data if the display were rendered as dB. The linear amplitude scale conveniently hides this very important weakness of poor S/N.

IQS Propaganda

The db magazine article of November 1983 by Jesse Klapholz is an excellent example of how useless the FFT approach is, especially when poorly implemented. The poor dynamics of the impulse response was shown in this article. The S/N would appear to be 20 dB or less. It was so poor that the peak amplitude was deliberately kept low on this (linear!!) display. Converting these to ETCs would be most embarrassing. The article erroneously refers to these figures as ETCs.

Another figure shows a very noisy phase response. This is most likely the direct result of poor S/N in the test as opposed to the too wide time window although it does its damage as well. Still another figure reveals the absurdity of computing the derivative of an incorrectly made phase plot as only the noise can be seen in the group delay. In order for group delay plots to be used, one must first be able to obtain a low noise phase plot.

The article shows that the aliasing filter is implemented with a switched capacitor filter which has only a single pole untuned aliasing filter preceding it and no aliasing filter following the switched capacitor filter. Switched capacitor filters require aliasing filters as they too are sampled data devices. A fully tuned second order analog filter should be used in both pre and post-filter positions. This is the approach of the TEF-10. Proto designs using a single pole fully tuned pre and post-filter system revealed aliasing errors in the switched capacitor filter. One may never know which noise signal they're watching in the dirt on the IQS display.

The IQS literature has an HP pen plotter output of a joint-domain display of apparently a speaker system. Little time resolution is evident in this display. The display shows good frequency resolution but poor time resolution. This is the nature of FFT systems to allow no control over the uncertainty tradeoff. Interestingly, this display is neither an Apple or an Epson display which is what the current product produces. People are led to believe that all they need do to get this cute plot of little use is to plop down their money.

There is a place for cheaply implemented equipment such as the IQS, but it is not in professional acoustic measuring environs. The proper place for such is in the classroom where a high S/N (textbook) environment can be produced. The FFT and its limitations should be readily demonstrable with this equipment. Defects such as poor bit resolution can be even taken as an advantage in the classroom as students must appreciate the limitations of physical implementations. The cost of the equipment is also reasonable for the classroom.

The limits of the Apple II's data storage capacity and storage integrity are also tolerable in the classroom setting. The low density controlerless disk implementation of the Apple has been observed to even tolerate data read errors without retry or error reporting and errors are frequent.

The unsuitability of the 6502 processor for digital signal processing (having no 16 bit hardware arithmetic instructions) is evident in the 18 seconds required to do a 1024 point FFT (and that on 8 bit data). The TEF-10 requires about .6 seconds to do the 1024 point FFT on 12 bit data and uses a math processor chip to achieve humanly tolerable response times.

There will no doubt be more products in the IQS vein to appear in the marketplace. The license exclusivity of TDS will assure that all are either patent violators (if they are useful instruments) or limited use devices in the realm of FFT processors. ♦

Rauland Field Report

ATTENTION: Kenneth T. James, Vice President, Sales/Marketing

SUBJECT: Equipment Failure or Malfunction

Date: ____________

Distributor: ____

Address: ____

City: ____________________ State: ______ Zip: ____________ Tele: (   ) ____________

Job Name: ______________________________ Contact: ____________________

Address: ____

City: ____________________ State: ______ Zip: ____________ Tele: (   ) ____________

Date of Installation: ____________________

The failure or malfunction reported below involves the following product line:

☐ TELECENTER ☐ DIRECTOR SERIES ☐ SPECTRUM-MASTER ☐ AMPLIFIER SERIES LINE

☐ RESPONDER ☐ PROGRAMMABLE CLOCK SYSTEMS ☐ OTHER ____________________

Product Model No. ____________________ Part No. Involved: ____________________

If part number is not known, please describe: ______________________________

____

Have you contacted your District Manager? ☐ YES ☐ NO

If yes, when was your District Manager informed? ______________________________

Have you spoken to anyone at Rauland about this problem? ☐ YES ☐ NO

If yes, to whom? ______________________________ When? ____________

Briefly describe the fault or malfunction: (Use the back of this form if necessary) ____________

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____

Reported by: ______________________________
(Please print)

Signed: ______________________________

We all have experienced the firms that once the item is delivered, lose all further interest in their product. Then there are the companies who feel that customers are adopted family members. Both these attitudes and all the gradations in between are the direct result of how *top* management really feels about their product, their reason for being in business in the first place, and their customers' intelligence and worth. That's why we were fascinated with the "field report" form shown here. Rauland, of course, does have a very close relationship with their carefully selected and supported distributors. We were pleased to see that these reports went to the man able to take action. The fewer hands such a report passes through, the better. We recommend the "thinking" behind this report to your attention. ♦

"HEARING-IT'S PSYCHOLOGY & PHYSIOLOGY"

The Acoustical Society of America has an extremely valuable program for the reprinting of hard-to-find source books in acoustics. Past reprints have included Hunts' *"Electroacoustics"*, Morse's *"Sound & Vibration"* and Knudsen and Harris' *"Architectural Acoustics."* The latest in this series is *"Hearing - It's Psychology and Physiology"* by Stanley Smith Stevens, Ph.D., and Hallowell Davis, M.D. The original book was published in 1938 and has firmly established itself as a classic in its field.

We would be hard put to find a dozen contemporary books that would allow the coordinated viewpoint this volume provides of the mechanisms, measurement, and models of the hearing process.

One of the aspects of such reprints is that it allows us to go back to the beginnings and it always surprises me to realize that yesterday's difficult is today's "we all know."

Under "the stereophonic effect" the authors describe the duplication of the Haas effect by making one channel predominantly reverberant and the other direct sound. "The virtual source was localized exactly at the 'direct' loudspeaker, until the power from the reverberant loudspeaker was from 8 to 10 dB greater."

A further quote leads to a fascinating modern experimental possibility. "In general, localization tends toward the channel giving the most natural reproduction..." This suggests to our thought, especially in the light of Puddie Rodgers' work - a pan pot research project utilizing a comb filter generator in conjunction with the usual intensity effects where the intensity effects automatically control the comb filter generator.

This paperback volume is well worth the modest asking price of $15.00. Send all orders to: Acoustical Society of America, 335 East 45 Street, New York, NY 10017. ♦

PASADENA - APRIL 12-13, 1984

WANTED: A Lock and Chain For Floppy Disks

Speaking of jobs created by computers, at least thieves of every stripe, from white collar to no collar, have seen the opportunities offered and seized them. The latest example, reported by the Associated Press, concerns a disk stolen from a booth at the convention of the American Institute of Architects in Phoenix, Ariz. It seems the $2.50 floppy contained computer-graphics data valued by its disconsolate owner at upwards of $2 million. All of which suggests still another can't-miss money-maker: design a lock and chain that can keep a floppy disk at its owner's computer.

Electronics/May 31, 1984 ♦

HE/SHE MAY BE WILLING TO

RENT HIS/HER ANALYZER AND HIS/HER EXPERTISE

You will note that two women are listed as TEF® owners. The heading for TEF owners in the 1984 winter issue of the Newsletter said, "These MEN are starting a Revolution." That was a mistake. Women are very much a part of the TEF revolution.

Mr. Ben Kok Peutz & Associates St. Ann Astraat 113 Nijmegen, 6524 EKN The Netherlands	Mr. Bob Grunberg Audio Supply P. O. Box 296 Double Bay, N.S.W. Australia 22023	Mr. Gordon Kuhl Alberta Audio Consultants 14524 118th Ave. Edmonton, Alberta T5L2M8 Canada	Mr. Hellmuth Kolbe Ing. Buro Fur Akustick Zielackerstrasse 6 Wallisellen CH-8304 Switzerland
Mr. Theodore Kowdrysh Box 127 Champlain La Salle H8P 3J1 PQ CANADA	Ms. Mary Gruszka 88 Myrtle Ave. Edgewater, NJ 07020	Mr. Butoni A T & T 50 Lawrence Road Springfield NJ 07091	Mr. Dave Andrews Andrews Audio Consultants 317 West 39th St. New York, NY 10018
Broadcast Technology 1619 Broadway, 10th Floor New York, NY 10019	Mr. Joe Silver Information Transmission 37-11 Crescent St. Long Island City, NY 11101	Mr. John Strohbeen Ohm Acoustics 241 Taafee Place Brooklyn, NY 11205	Mr. Charles Biliello 258 Fairlawn Ave. West Hempstead, NY 11552
Mr. Phillip Clark Diversified Concepts, Inc. 3929 New Seneca Turnpike Marcellus, NY 13152	Mr. Ron Borthwick Clair Brothers P. O. Box 396 Lititz, PA 17543	Mr. William Gellhaus WRMG Studios P. O. Box 73 Cheltenham, PA 19012	Mr. Bruce Howze Community Light & Sound 333 East Fifth St. Chester, PA 19013
Mr. Emile Mallette Nat'l Bureau of Standards Rm A825 Administration Bldg Washington, DC 20234	Ms. Gina Becker Audio Artistry 10120 Ashwood Drive Kensington, MD 20895	Mr. W. Peterson Professional Sounds, Inc. 709-B West Broadway Falls Church, VA 22046	Mr. Mike Hoover Audio Technical Services 239 Mill St. N.E. Vienna, VA 22180
Mr. Richard Clark American Multimedia P. O. Box 2154 Burlington, NC 27215	Dr. Eugene Patronis Georgia Tech/Dept of Physics 888 Hemphill Ave. N.W. Atlanta, GA 30332	Mr. Richard Lee Richard Lee Associates 1755 N.E. 149th St. Miami, FL 33181	Mr. James Carey James Carey Associates 1710 Lowry Ave. Lakeland, FL 33801
Mr. Robert Todrank Valley Audio 2821 Erica Place Nashville, TN 37204	Mr. John Murray Panacom Corporation 2230 N. Main Street Dayton, OH 45405	Mr. Glen Meeks Comcast 1253 S. Shepard Indianapolis, IN 46221	Mr. Sandy Swartzendruber Precision Audio, Inc. 18582 U. S. 20 Bristol, IN 46507
Mr. Henry Root Hy James, Inc. 2839 Boardwalk Ann Arbor, MI 48104	Mr. Harold Helse Ford Motor Company P. O. Box 1704 Dearborn, MI 48121	Mr. Paul McGuire Electrovoice 600 Cecil St. Buchanan, MI 49107	Mr. Mark Doubet WMT Music & Sound 796 11th St. Marion, IA 52301
Mr. Richard Buesing Location Electronics 42 N. Linden Avenue Palatine IL 60067	Mr. John G. Mitchell 1120 Stonehedge Dr. Schaumberg, IL 60194	Mr. M. Langer Shure Brothers 222 Hartrey Ave. Evanston, IL 60204	Jay Bridgewater Bridgewater Custom Sound 15957 S. Halsted Harvey IL 60426
Mr. Garrott Elghammer Garrett Sound 664 N. Michigan Ave. Chicago, IL 60611	Mr. James Brown 936 West Montana Chicago, IL 60614	Mr. Brock Jabara Galaxy Audio 625 East Pawnee Wichita, KS 67211	Mr. Harvey Earp J. W. Davis & Co. P. O. Box 26177 Dallas, TX 75226
Mr. Russ Berger Joiner-Pelton-Rose 4125 Centurian Way Dallas, TX 75234	Mr. Jack Maxon Showco 9011 Governor's Row Dallas, TX 75237	Mr. J. T. King Audiometrics P. O. Box 3263 Longview, TX 75602	Mr. Pete Griffin Radio Shack 1100 One Tandy Center Fort Worth, TX 76102
Mr. Rob H. McKinley, Jr. L D Systems, Inc. 467 West 38th St. Houston, TX 77018	Mr. Roland Brazier Taft Broadcasting Corp. 4808 San Felipe Rd. Houston, TX 77056	Mr. Steven Hodge Box C8 College Station, TX 77844	Mr. Bob Herrick Production Consultants 642-A West Rhapsody San Antonio, TX 78216
Mr. Deward Timothy Poll Sound 3423 S. W. Temple Salt Lake City, UT 84115	Mr. Bob Crain 5320 Constitution N.E. Albuquerque, NM 87110	Mr. John Bau Spica 1601 Paseo De Peralta Santa Fe, NM 87501	Mr. Chips Davis LEDE Designs 3364 Clandara Ave. Las Vegas, NV 89121
Mr. David Jensen 413 Gomez Court Sparks, NV 89431	Mr. Ludwig Sepmeyer 1862 Comstock Ave. Los Angeles, CA 90025	Perception, Inc. 1537 Cerro Gordo St. Los Angeles, CA 90026	Mr. Jerry Yokum Douglas Aircraft 2401 East Wardlow Rd. Long Beach, CA 90807
Mr. John Prohs Ambassador College 150 West Dayton Pasadena, CA 91123	Mr. Rollins Brook Bolt, Beranek & Newman P. O. Box 633 Canoga Park, CA 91305	Mr. Jim McCandliss 31131 Via Collins 1003 Westlake, CA 91361	Mr. Jeff Wimsatt Comdial 2066 W. 11th St. Upland, CA 91786
Mr. Joe Martinson Martin Sound 1151 W. Valley Blvd. Alhambra, CA 91803	Mr. Bill Lipis Sound Maintenance Co. P. O. Box 325 El Cajon, CA 92027	Mr. Don Davis Syn-Aud-Con P. O. Box 669 San Juan Capistrano, CA 92693	Mr. Algis Renkus Emilar Corporation 1365 North McCan Street Anaheim CA 92806
Mr. Richard Vandersteln Vandersteln Audio 116 W. 4th St. Hanford, CA 93236	Mr. Jeff Long P F Sound #101 2727 N Grove Industrial Dr Fresno, CA 93727	Mr. Niles Christenson 2600 Garden Road #125 Monterey, CA 93940	Dr. Don C. Creevy 1175 Westridge Drive Portola Valley, CA 9402
Mr. Timothy Purcell 3020 Balboa St. San Francisco, CA 94121	Mr. Kean Sakata 2814 Dohr St. Berkeley, CA 94702	Mr. Randy Bauske 1684 Kalakaua Ave. Honolulu, HI 96815	Mr. Mike Garrison H I S Sound 715 S.E. Grand Ave. Portland, OR 97214

AES PRESIDENTIAL CANDIDATE

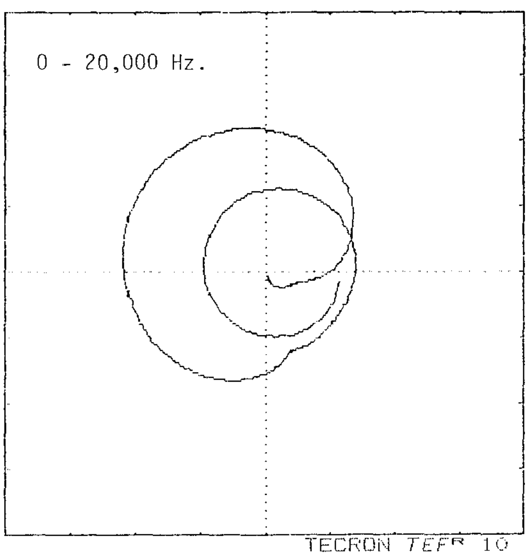
Robert B. Schulein is on the AES ballot this year for President Elect. Syn-Aud-Con hopes that all Syn-Aud-Con graduates will support this fellow graduate with your votes. We are pleased to see a qualified representative of an American manufacturing firm in the running for this office.

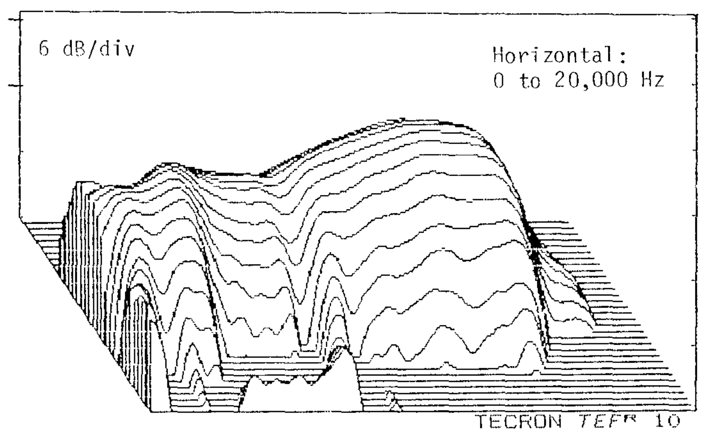
Robert Schulein talking with Don, Dr. Patronis (left in photo) and Gerald Stanley (right).

Bob is Chief Development Engineer of Shure Brothers Incorporated in Evanston, Illinois, where he is responsible for research and development activities in the areas of microphones, phonograph cartridges, and loudspeaker systems. He is a Fellow of the AES and a member of the Editorial Board of the AES Journal. In 1977, Bob received the AES Publication Award for the outstanding paper published in the AES Journal during the two preceding years -- "In Situ Measurement and Equalization of Sound Reproduction Systems." ♦

SONY HI-FI SPEAKER

Ken Gruber, Supervisor of the Sound Department at Disneyland, Anaheim, CA, was one of the supervisors of the sound system for the Disneyland/Tokyo Bay project. He brought back a Sony hi-fi loudspeaker. He especially liked the sound and asked us if we would measure it during the evening after the Anaheim class.





Note the smoothness of the Nyquist and the outstanding backside of the waterfall. ♦

OBTAINING A BESSEL ARRAY NETWORK

The J. W. Davis and Co. offers a small inexpensive ($99.50) five speaker column, the DB-5 Be-Ray. The network used in this unit can be employed to make any five loudspeakers into a Bessel array, i.e., five loudspeakers with the polar response of one loudspeaker of the same type. A twenty-five loudspeaker Bessel array can be formed from six of these networks. Those of you who have seen our demonstrations of these units in our recent classes now have an economical way to become more familiar with them. ♦

POWER AS A FUNCTION OF LOAD IMPEDANCE

Farrel Becker's chart is extra useful because of the choice of parameters he used to make the plot. By choosing an E_S = 10 volts and an R_S = 10 ohms, the left hand vertical scale can be read as P_L in watts for the curve using □, as R_L in volts for the curve using +, and as I_L for the curve using ◇. This is because:

$$E_L = \frac{E_S\,R_L}{R_S+R_L} = \sqrt{WR}$$

$$I_L = \sqrt{\frac{W}{R}}$$

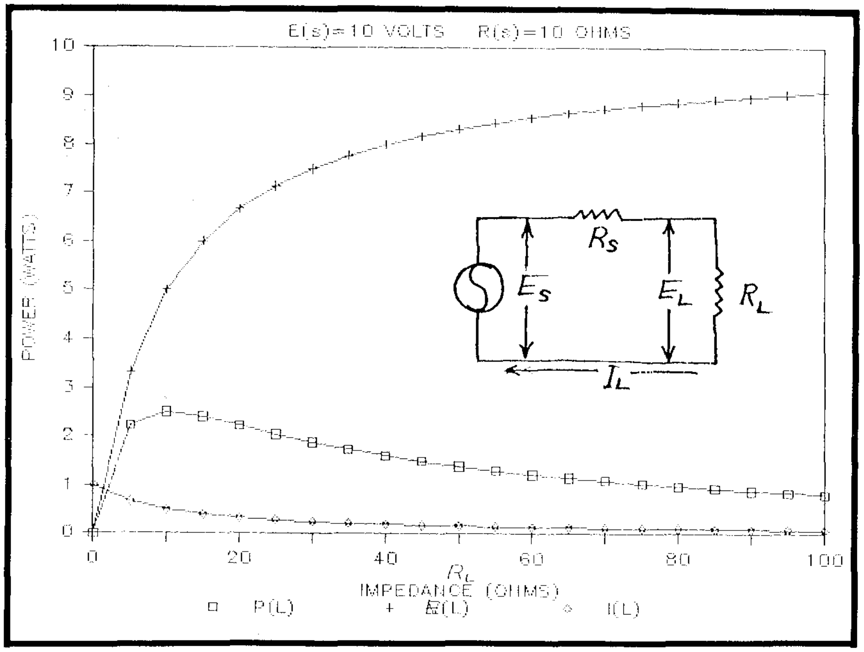
and: $$P_L = \frac{E_L}{R_L}$$

Example:

For an R_L = 10Ω $E_L = \left(\frac{10(10)}{20}\right)$ = 5v

$$I_L = \frac{2.5w}{10} = 0.25 \text{ amp}$$

$$P_L = \frac{(5)^2}{10} = 2.5 \text{ w}$$ ♦



AUDIO EDUCATION

We are encountering increasing rhetoric from within the AES that they should take a more active part in education. We agree!

The trouble is that, so far, the approaches suggested are ego trips for selected cliques within the political side of the organization. The way for any peer group technical society to effectively support specialized education is to *provide funds* to qualified already existing educational facilities.

Suggestions include funding dedicated graduate students under the supervision of known audio authorities, such as Doctors Patronis, Greiner or Pries. Certainly, Al Grundy's Institute of Audio Research has more than proven its worth over the years.

We sincerely hope that the AES has no more hype oriented conferences in mind under the disguise of audio education but staffed by product oriented, manufacturer associated personnel.

We were interested to read an editorial in *Consulting Engineer* that related how the HVAC professionals are working with both manufacturers and related technical societies to obtain funding *for qualified educational establishments*. The AES and ASA should take note. ♦

Need HVAC Engineers? Read This

There is little doubt that there is a serious shortage of engineers trained in heating, ventilation, and air conditioning. The main reason appears to be the lack of HVAC courses offered by the mechanical engineering departments at the universities. Working on the premise that the architectural-engineering departments, through their environmental options, would be a logical source of engineers trained in this discipline, Lou Bacon, President of the National Society of Professional Engineers, brought together the heads of the eight accredited A-E departments, a representative of ASHRAE, six other HVAC consultants, and two gas utility company officials for a brainstorming session. The American Gas Association sponsored the one-day session.

Both the academics and the potential employers profited. The fact that there are A-E programs at eight universities came as a surprise to most of the consultants who had been seeking graduates of mechanical engineering departments at engineering schools, where there is a general lack of interest in the field of HVAC by faculty. The professors, some whose schools were seriously considering dropping HVAC from the curriculum, came away with a better grasp of the need for these graduates.

The group reached several conclusions: most A-E programs are structures-oriented because they lack faculty to teach HVAC and the schools perceive a lack of demand for these students because HVAC firms do little or no on-campus recruiting.

This led to four recommendations for action, all of which involve consulting engineers and their associations:

- The professional societies need to alert their members to the existence of the eight A-E programs.
- The professions need to contact the schools by letter and personal visit to encourage the continuance of the A-E programs and the HVAC option.
- The professional societies should develop a program to generate funds from the utilities, HVAC equipment manufacturers, and consultants for research and to augment professors' salaries so the schools can hire experienced teachers, at least during the initial years of the rebuilding program.
- The three professional societies — NSPE, ACEC, and ASHRAE — should organize teams of HVAC engineers to visit the schools to demonstrate the need for the HVAC option.

If your firm is interested in encouraging more young people to enter the HVAC field, this is your chance to do something positive about it. Call your society to ask how you can participate or contact Louis Bacon, NSPE, 2029 K Street, N.W., Washington, DC 20006.

"I TOLD YOU SO"

It's not polite to say "I told you so," but I'm tempted beyond my ability to resist. Next question. ♦

FAULT IS IN THE FILTERS

Researchers Identify Cause Of Digital Audio Flaws

Special to AUDIO TIMES

PARIS—A remedy seems to be at hand for the sonic irregularities some claim to hear in digital recordings. In fact, the groundwork for such a solution has been around at least since the 1940s.

In a paper delivered here during the Audio Engineering Society convention in

AES CONVENTION

late March, Dr. Roger Lagadec of Switzerland's Studer described joint research by his firm and U.S.-based Soundstream, identifying the degeneration of audio signals when converted from digital to analog format. Both companies have been long involved in research and manufacturing for digital recording systems.

According to Lagadec, the "dispersion" of the signal—time and amplitude changes—occurs in the filters used to convert a digital signal to analog. Using a 400 Hz tone, familiar to professional recording engineers for its gentle attack and slow decay, the researchers processed the signal through a theoretically-correct 16-bit filter. However, the tone was audibly different when it re-emerged in analog.

Lagadec contended that the "dispersion" effects, which do not take place in analog processing, also occur at other frequencies. Recording engineers and other "golden ears" often have difficulty describing the signal alteration, he noted, because no equivalent exists in analog.

"In going from analog to digital, it seems we have broken some implicit rule of analog processing," said Lagadec. "If I had to bet as a scientist, the way digital audio is done today we have accepted effects that must be audible to 'golden ears'. If we want to do the best we can with 16-bit digital-to-analog processing, we have to take these dispersion effects into consideration very seriously."

Hurdles Overcome

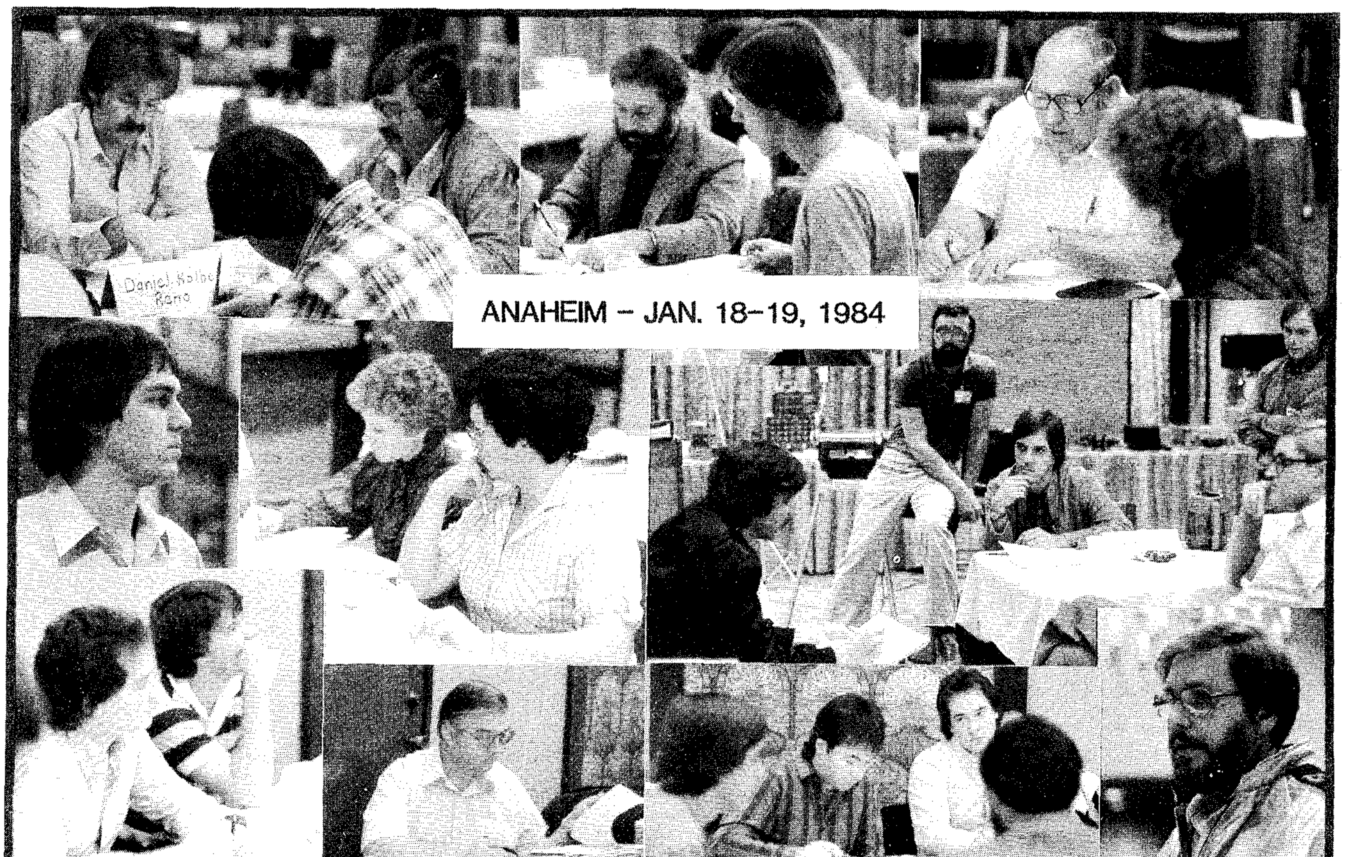
Lagadec explained that these hurdles can be overcome. The irregularities, he said, can be predicted now that engineers understand what they're looking for.

The remedy, he noted, is simple—though it will take time to develop and build the proper filters. According to Lagadec, descriptions of how to make dispersion-free filters exist in the public domain, in the form of 1939 research for telephony. References to the problem can be found in 1929 Bell Labs research—and earlier.

"It is prior art," said Lagadec. "We have no excuse."

In other AES discussions, Studer presented a low-cost, degeneration-free means of electronic editing for studio digital recorders. And Mitsubishi described a consumer digital recorder that uses metal-evaporated Compact Cassettes.□

AUDIO TIMES May 1984



READING SIMPLE ETC DISPLAYS

Before any measurement is undertaken, it is wise to be aware of what should happen so when a distinctly different thing occurs, it is recognized as a significant difference.

A Real Life Example

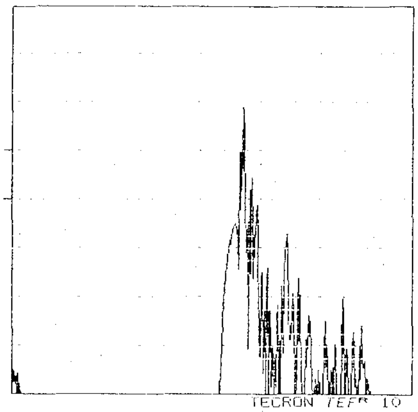


Fig. 1: An "Unaligned" Monitor Loudspeaker

Vertical: 6dB/div with base of display at 42.1dB
0dB is located at .00002 Pascals

Horizontal: 0 microseconds or 0 Feet to
6278 microseconds or 7.09364 Feet
scale: 1.9395E+00 Feet/inch or 7.6357E-01 Feet/cm.
1716 microseconds/inch or 675 microseconds/cm.

Line Spacing: 15.7332 microseconds or 1.77785E-2 Feet
Line Width: 21.3971 microseconds or 2.41787E-2 Feet

Sweep rate: 5009.55Hz/Sec

Sweep range: -31778.90Hz to 31781.00Hz

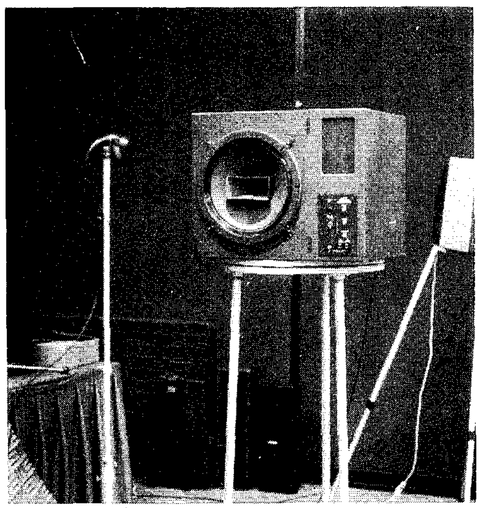
In a recent class some friends brought in a very early UREI 811 monitor that had a non "Time Align*" network. The ETC shown in Figure #1 is for the whole loudspeaker system. Note that we are using a signal swept from a -31778 Hz to a +31781 Hz; that is, we sweep from -31778 Hz downward until we reach 0 Hz and then continue on upward to +31781 Hz (a la Charles Bilello).

What Are Negative Frequencies?

What are negative frequencies? They start from 31,778 Hz with a different polarity than do positive frequencies. The sweep is downward to zero Hz, where it reverses sign, and then upward again to 31,781 Hz. This provides a total bandwidth of

$$31{,}781 + 31{,}778 = 63{,}559 \text{ Hz.}$$

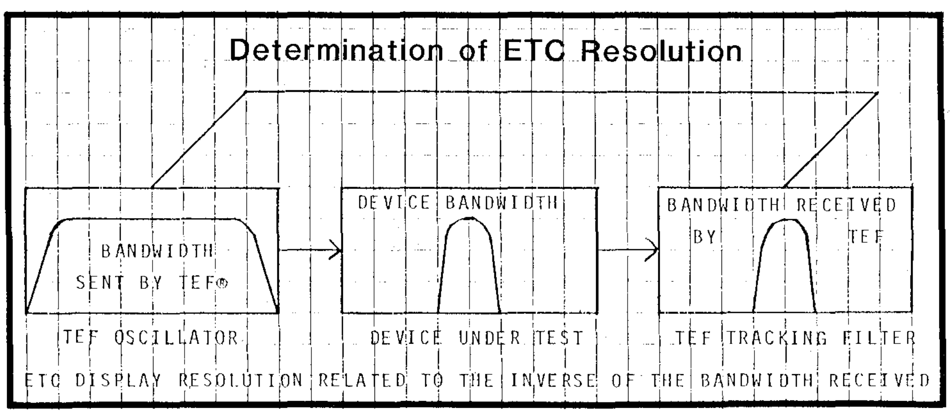
1/63,559 = 0.000015733 secs, which is the line spacing for the 400 lines constituting the ETC display. If we multiply the line spacing by 1.36 (a filter weighting factor), we obtain the line width of 21.3971 usecs (microseconds). Multiplying 400 times the line spacing gives the full screen time 6.28 msecs (milliseconds).



Loudspeaker under test.

The "Frequency Blind" TEF??

Technically, an ETC display is frequency blind. In actual practice some frequency data can be gathered from the ETC display. The swept frequency mode is a linear sweep. This means that the height of the time line of a perfect device would be divided into equal height per Hz. Therefore, 1000 Hz to 10,000 Hz will be 10 times the height of 0 to 1000 Hz. TEF® users have also noted that the lower frequency components are *broader in time* than the higher frequency components.



A fruitful area of investigation would be to carefully measure what bandwidth is necessary to produce a given width ETC display. Suggested experiment:

Obtain a high quality bandpass filter set, perhaps a sixth order unit. Carefully note time widths vs bandwidths and once this correlation is obtained, investigate how various reshapings of the frequency response within that bandpass affect the time width of the ETC display.

*Trademark of E.M.Long Associates.

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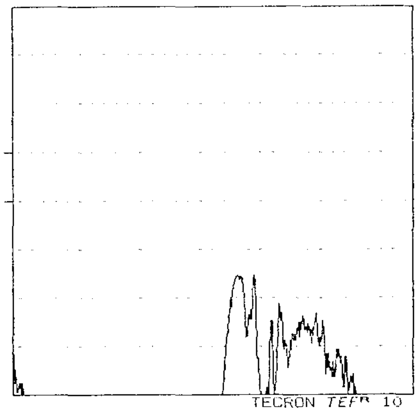


Fig. 2: Woofer Only of a Monitor Loudspeaker

```
Vertical:  6dB/div with base of display at   42.1dB
           0dB is located at  .00002 Pascals

Horizontal:        0 microseconds or 0 Feet to
                6278 microseconds or  7.09364 Feet
     scale:  1.9395E+00 Feet/inch or  7.6357E-01 Feet/cm.
             1716 microseconds/inch or  675 microseconds/cm.

Line Spacing: 15.7332 microseconds or  1.77785E-2 Feet
Line Width:   21.3971 microseconds or  2.41787E-2 Feet

Sweep rate:  5009.55Hz/Sec

Sweep range: -31778.90Hz to  31781.00Hz
```

Now, let's go back to our ETCs of this particular monitor loudspeaker. If the woofer can only pass 40 to 1500 Hz, then no matter how wide our TEF output sweep, the TEF input will see only the 40 to 1500 Hz bandwidth which then becomes the time resolution limit. The inverse of this T_R is correlated to the bandwidth the device passed. In this particular case, if we assume the bandwidth given above, the T_W at the base line of the woofer ETC is very close to the inverse of the bandwidth.

The woofer ETC is the result of covering the driver of the high frequency horn with a piece of Sonex while running the ETC measurement. With some preknowledge about the device we are measuring, such as its approximate crossover frequency, we can then easily identify which parts of the ETC are the results of which part of the total frequency response. Note carefully, however, that a restriction *of the bandpass* is what caused the widening of the display and that restriction could have occurred at either end of the total possible bandpass. ♦

Loudspeaker under test.

SEATTLE – MARCH 21–22, 1984

TIME AND DISTANCE INVESTIGATIONS

Looking at the two classic chartings of sound fields, Figs. 1 and 2, we find that the first contains the coordinates of level vs. distance, the second level vs. time. It occurs to us that several very worthwhile investigations can be undertaken with TEF® analysis.

1. What is the equation for the time requirec for the reverberant sound field to develop? Is it the same as the decay time?
2. What is the equation for the natural room delay?
3. The timing associatec with initial time delay gaps and the mean free paths.

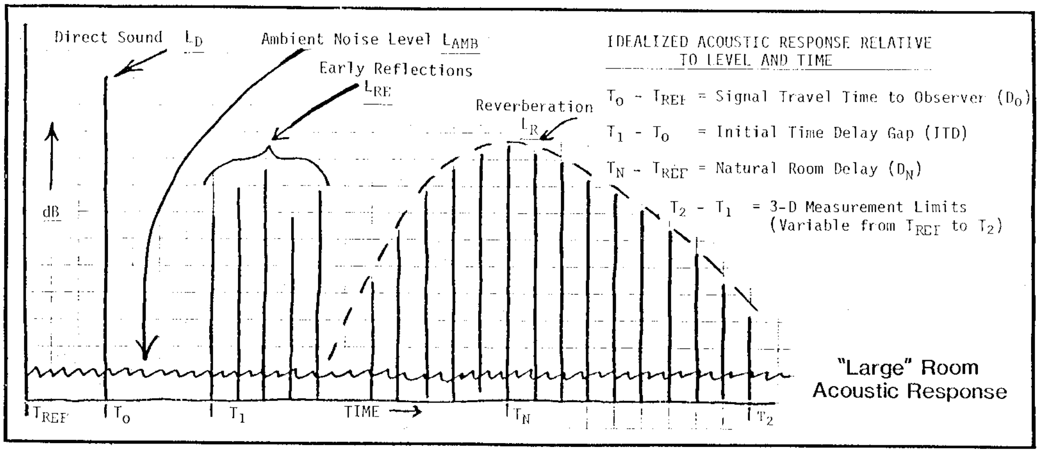


Fig. No. 1

Acoustic Level vs Distance

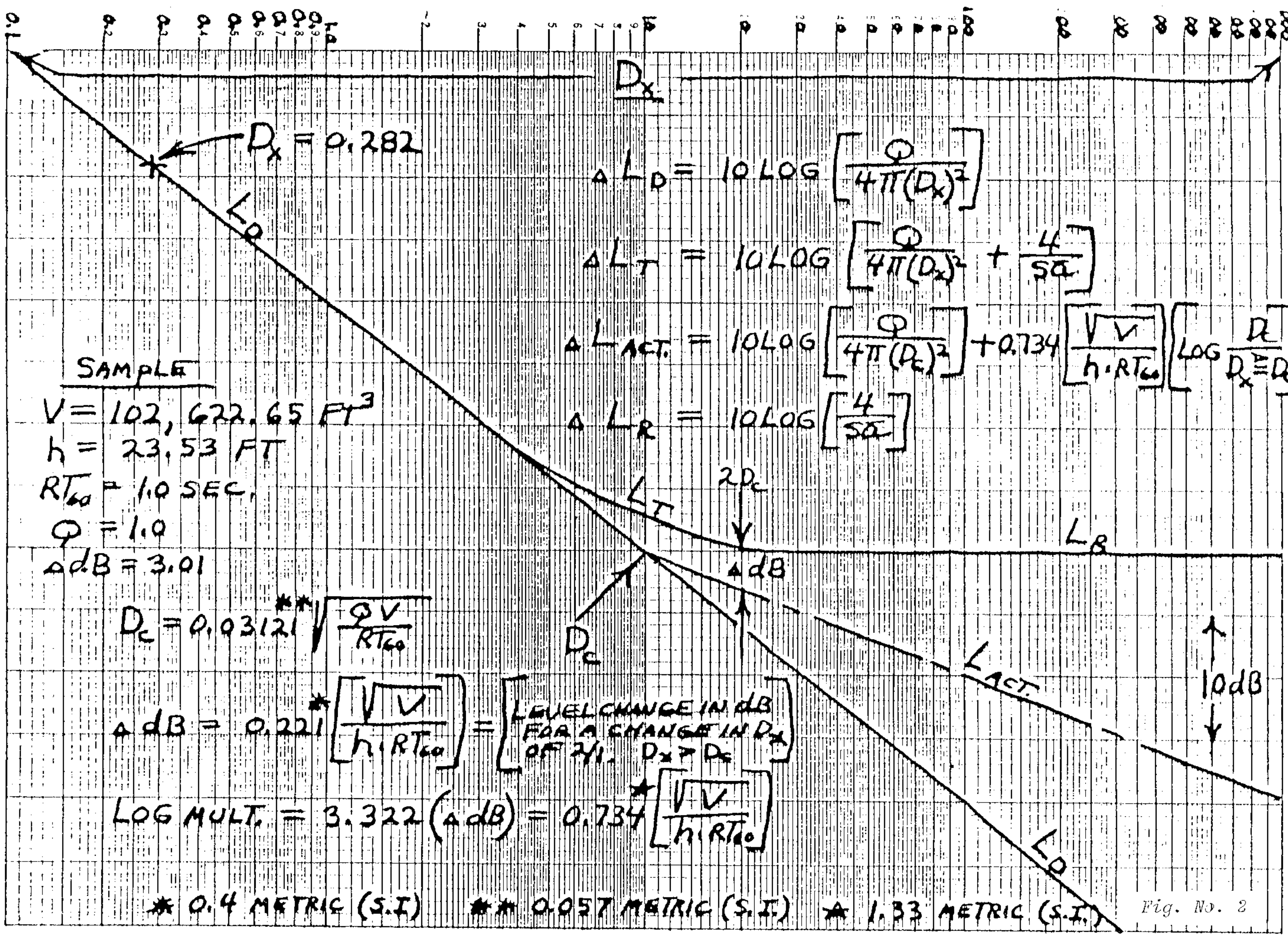


Fig. No. 2

MANIPULATING L_W IN ARRAY DESIGN

Dr. Eugene Patronis and David Harris have brought to our attention a special case utilizing the manipulation of L_W in array design.

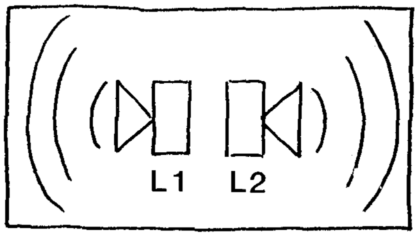
Some Preliminary Considerations

Outdoors, or when indoor conditions essentially approach "free field" conditions, i.e., ΔdB = -5 or greater, the level of L_D can be adjusted by changing L_W instead of Q. In highly reverberant spaces, ΔdB = -1 or less, adjusting L_W results in one device controlling the entire reverberant sound field, L_R, and experience has shown that adjustment of L_D should be done by varying the Q of the device covering the area where it is desired to change L_D.

The Semi-Reverberant Case

In the semi-reverberant case, where ΔdB falls between -1 and -5dB, some manipulation of L_W in an attempt to control L_D might be acceptable. The concept is that (10 LOG_N) + L_W remain a constant while the L_W of the devices are manipulated.

MANIPULATING L_W IN ARRAY DESIGN



L_D depends on L_W, Q, D_X, Me
L_R depends on L_W, Sā, Ma, N

Case Number One

L1 has an L_W = 123 dB
L2 has an L_W = 123 dB
L_T has an L_W = 126 dB*

Let $L_D - L_R$ = -6 dB in front of L1 and L2

Case Number Two

L1 has an L_W = 125 dB
L2 has an L_W = 119 dB
L_T has an L_W = 126 dB*

$L_D - L_R$ = -4 dB in front of L1 but is now -10 dB in front of L2

*Total L_W = (10 LOG N) + (L_W of device producing L_D)

In the first case shown in the illustration N = 2 and L_W = 123 dB. Thus (10 LOG 2) + 123 = 126 dB.

In case number two: $$N = \frac{(Wa_1\ We_1\ Ut_1) + (Wa_2 + We_2 + Ut_2)}{Wa_1\ We_1\ Ut_1}$$

Therefore, N for a listener in front of L1 becomes: $10^{\left(\frac{.119}{10}\right)}$ - (10 LOG 1) x 1 x 1 = 0:82 and the ratio of $\frac{2}{0.82}$ becomes a level change of: $10 \text{ LOG} \left(\frac{2}{0.82}\right)$ = 3.87 dB or L_D rises approximately +4 dB in front of L1 while the L_R remains the same.

Note, however, that the L_D-L_R in front of L2 is now -10 dB instead of -6 dB. This would suggest either that it would be just as well to turn L2 off or that any listener would have to sit much closer to this source.

Syn-Aud-Con prefers to limit the measurement of Q to individual directional control devices and to use the N parameter to account for the behavior of multi driver complex arrays. With the advent of true *intensity* analyzers, it is conceivable that new techniques may be introduced that attempt the measurement of complex array Q. If that is done, the computational techniques will have to be upgraded as well. The present techniques allow simple tools for both calculation and measurement and provide a notation system minus redundant use of the same symbol for two different parameters. ♦

WESTINGHOUSE GROUP

Syn-Aud-Con had the privilege, just before NAB in Las Vegas, to have a one-day special seminar for Westinghouse broadcast engineers.

We were pleased to realize their growing need of the TEF® type analysis in their broadcast work as stereophonic sound is increasingly used. The increased awareness of the role of psychoacoustics in utilizing signal processors makes increased access to high resolution time domain measurements a vital tool. ♦

ANECHOIC CHAMBERS

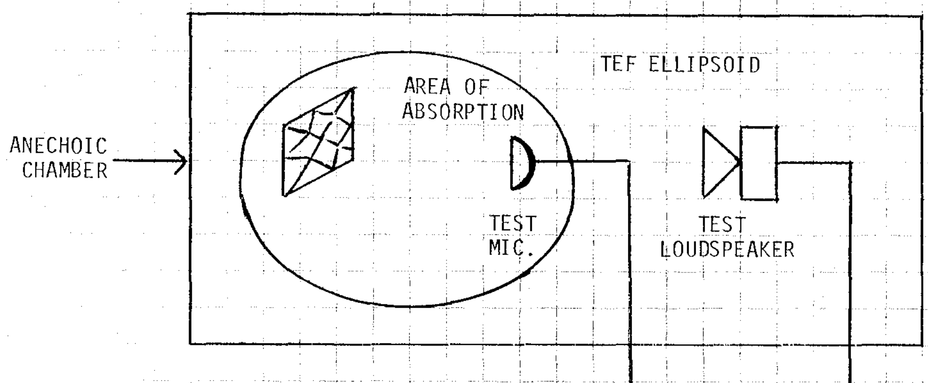
As we discuss TEF® measurements with engineers who normally utilize anechoic chambers, we find that they sometimes do not have a grasp on the constraints such a chamber imposes.

Low Frequency Measurements

For instance, they view the TEF ellipsoid as if it were a constraint unique to the TEF analyzer. It is, of course, a constraint on any measurement technique. Since the ellipsoid only has to be large when low frequency resolution is desired and anechoic chambers do not have adequate absorption at low frequencies (be assured none do), the constraint becomes, in the TEF ellipsoid case, a constraint on the distance to the nearest surface of *any type* whereas the anechoic chamber constraint becomes that of reflective interference with the desired signal below a certain frequency. With the properly chosen TEF ellipsoid (and we might add realistic size in the world of real buildings available to everyone) we have precise inverse square law level changes down to any low frequency we choose. (See Fig. 1) *Most anechoic chambers have ceased such behavior in the region of 100 to 200 Hz.*

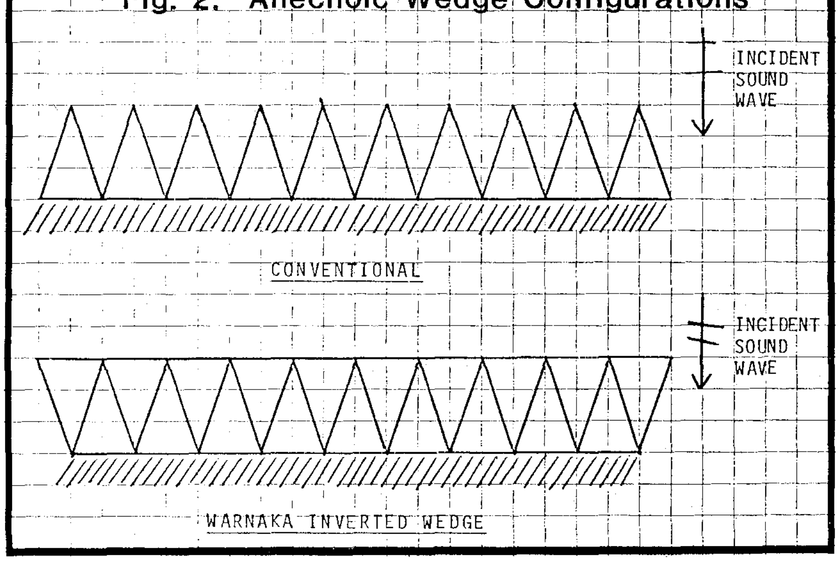
In the case of higher frequencies, really good anechoic chambers will reduce unwanted reflections by 20 dB. Signals outside the TEF ellipsoid are discriminated against by 60 dB.

Fig. 1: Patronis' Method of Measuring Reflections



1. Choose ellipsoid on basis of 1/f, sample size, or both.
2. Put selected area for measurement inside ellipsoid.
3. Put measurement microphone inside ellipsoid.
4. Put test loudspeaker outside ellipsoid.
5. Measure reflection level relative to non-absorptive surface of equal area.

Fig. 2: Anechoic Wedge Configurations



S/N Ratio

One of the tertiary uses of an anechoic chamber is the isolation such a structure provides from undesired ambient noise sources.

The TEF analyzer's narrow receiving filter sees only a narrow bandwidth of the noise but all of the synchronized test signal, thereby yielding a 20,000 to one S/N advantage over a conventional analyzer. It is normal for us to utilize the entire 72 dB dynamic range of the analyzer, even in the presence of ambient noise, without seeing the noise or its influence ever appear in the display. In the TEF case, the sound level meter's electronic noise floor is the limiting factor, not the acoustic environment.

Reflections Off Wedges

A very creative use of a TEF analyzer is to measure just how bad an anechoic chamber really is by measuring the reflections off the absorptive wedges, wire floors, speaker mounting fixtures, etc. (See Fig. 1)

Wedges Turned Wrong Way

As if all of the above were not sufficiently disturbing to engineers saddled with obsolete tools, Glenn E. Warnaka published a paper in the *Journal of the Acoustical Society of America* demonstrating that all anechoic chambers are constructed with *the wedges turned the wrong way*. (See Fig. 2) His company, Lord Corporation, has filed for a patent on the correct way. (See JASA 75(3), March 1984, pp 855-858) ♦

KEN WAHRENBROCK

We're often asked in our classes, "How's Ken Wahrenbrock and what is he doing these days?".

One of the things he does is to drop in on any and all Syn-Aud-Con classes in the Los Angeles area as well as anywhere there is a class and he happens to be traveling in the vicinity. Ken helped us with our Pasadena class this spring. He is showing the class a few of his new multi-boundary microphones.

Ken gets many calls from people with special microphone needs. He never asks, "What's in it for me?" He goes to his garage and starts a new boundary design. Ken is not limited by a wealth of advanced microphone theory, so he will try anything. (Syn-Aud-Con Tech Topic Vol. 11, No. 4, by Ken shows his latest work.)

Therein lies the great future for PZM. As a few more inventive, inquisitive "what happens if" people mold the plastic into different shapes -- shapes and forms that conventional microphony cannot assume -- the pressure zone concept will have broader and broader applications. ♦

A PRODUCT YOU SHOULD KNOW ABOUT

Dear Carolyn & Don:

In response to Mary Gruszka's comments, I would like to offer the following:

1. Regarding the VU/PEAK same time display, Mary wanted peak information for headroom protection. The uniVUer has an independently adjustable PEAK FLASHER. It may be adjusted to satisfy this need. Regarding the VU/PPM side by side, the carefully defined PPM graticule standards make this impossible. We have provided customers with 1-channel VU and 1-channel quasi-peak, but we are very careful to avoid the common error of applying the term PPM to anything other than EBU 3205-E.

2. Calibration numbers have been intentionally left off the uniVUer. Under most situations, the size of the display on the monitor will make reasonably sized numbers only a half dozen or so scan lines high, virtually unreadable. The graticule marks are chosen to replicate the standard VU or PPM markings. We have included a copy of the scale with numbers added.

3. The uniVUer is available with optional passive video bypass relays in the rackframe. This bypasses video if the rackframe power supply should fail or when the uniVUer card is pulled out. The option costs $250 per rackframe.

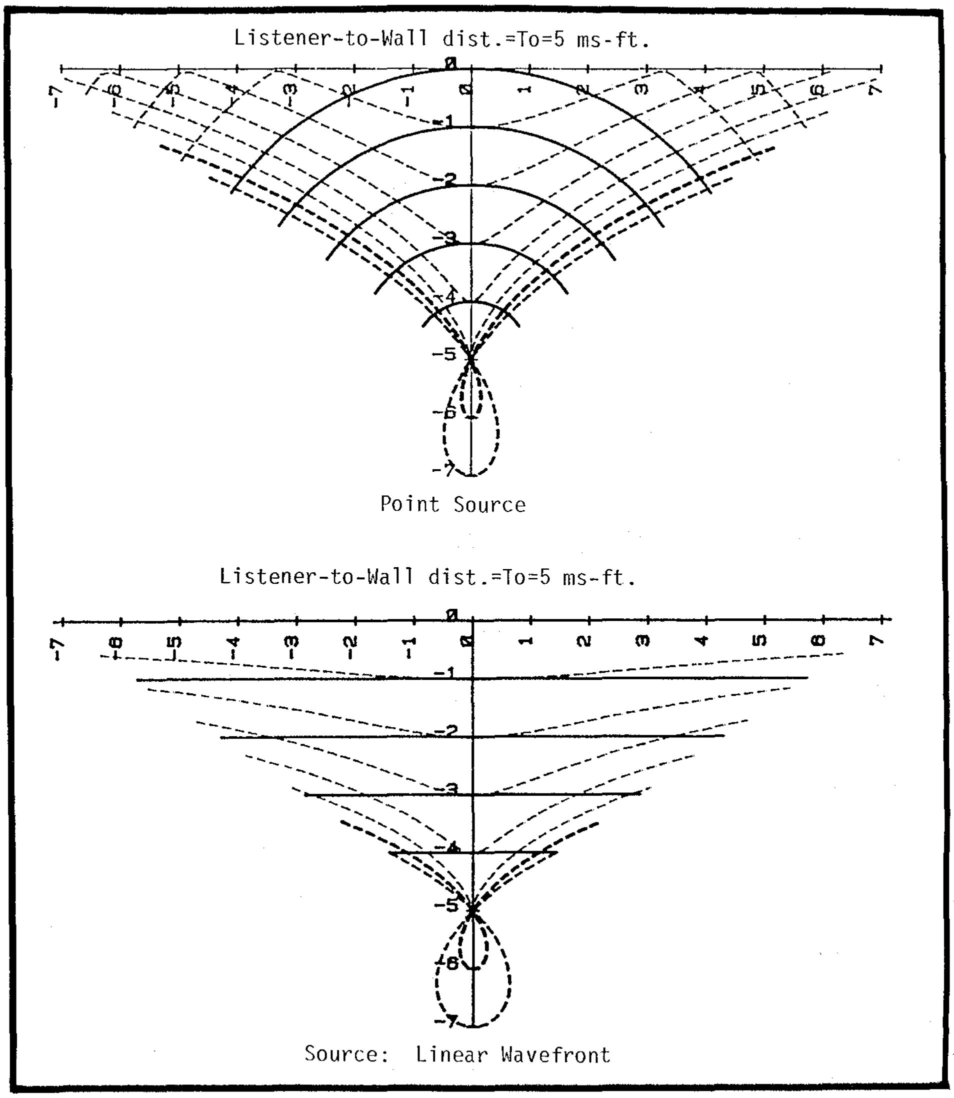
Incidentally, all three networks have evaluated the uniVUer in their labs in New York, and all three have approved it for use. ABC put 24 uniVUer PPM units into their control room at the Winter Olympics facility in Yugoslavia.

Regards,

s/Ray Terrell, President ♦

WALL REFLECTIONS

Joe Mitchell has once again plotted a set of curves of value to all of us seeking an intuitive sense of how sound energy behaves under various constraints. A beautifully conceived and executed example of wall reflections.



When a sound pressure wave strikes a surface, it dissipates in all directions including the direction from which it came. Just so, every point on the surface reflects sound back to a point located near the surface.

Here is a plot of wavefronts of sound striking a wall and reflecting back to a point located T time from the wall at intervals of

$$\frac{1T}{5}\quad \frac{2T}{5}\quad \frac{3T}{5} \ldots 2\ \frac{2T}{5}\ \text{etc.}$$

A) from the point (shown circular, but are spherical)
B) from infinite distance (parallel wavefronts)

Notice that if T is less than or equal to about 10 milliseconds, the direct sound will be followed 2T seconds later with a time smeared reflection, masked totally to the human ear! (Haas Effect) ♦

WALL REFLECTIONS

```
0: pclr
1: dsp "Lstr-to-
 wall dist.=To
 in ms-ft=5";
 5→T;fxd 0;stp
2: ent "+,- Deg.
  Deviation from
  the Normal?",D
3: scl -9.5,9.5,
 -10.5,2
4: xax 0,1,-7,7,
 1
5: yax 0,1,-7,0,
 1
6: plt 0,.5,0;
 cplt -18,0;lbl
 "Listener-to-
 Wall dist.=To=
 5 ms-ft"
7: ent "Deg.
 Increment=?",I
8: ent "Paralell
  or List.→Wall?
   1 or 0",P
9: plt 0,-8.5,0;
 if P=0;cplt -6,
 0;lbl "Point
 Source";jmp 2
10: cplt -12,0;
 lbl "Source:Lin
 ear Wavefront"
11: enp "Enter
 a,a(To)",A;if
 A>1;ent "pen#?"
 ,Q;pen# Q;line
 2,1
12: 90+D→C;90-
 D→E
13: for B=C to
 E by -I
14: if P;if A>1;
 -(A-1)Tsin(B)→Y
 ;Tcos(B)/sin(B)
 -(A-1)Tcos(B)→X
 ;jmp 4
15: if P;if A<1
 or A=1;ATcos(B)
 /sin(B)→X;(A-
 1)T→Y;jmp 3
16: ATcos(B)→X;
 ATsin(B)-T→Y;
 if Y=0 or Y<0;
 jmp 2
17: 2Tcos(B)/
 sin(B)-X→X;-Y→Y
18: plt X,Y
19: next B
20: pen
21: plt 4,1,1
22: gto 11
23: end
*6965
```

SMILE

FELSON'S LAW: To steal ideas from one person is plagiarism; to steal from many is research. ♦

HIRE THE RIGHT INSTALLER

Alpha Audio (of Sonex fame) in Richmond, Virginia, publishes a newsletter. Each newsletter is a powerful sales tool for Alpha Audio. Any company that has the time, talent and money should consider some form of printed communication with their customers and potential customers.

Nick Colleran, President of Alpha Audio, gave us permission to reproduce "How to Hire the Right Installer."

AlphaAudio

Prolines

HOW TO HIRE THE RIGHT INSTALLER

Getting the right firm to handle your installation should be a matter of common sense; but in that it involves technical stuff, and not all persons desiring a state-of-the-art system as technical people, perhaps a little coaching from the pros might help.

Alpha Recording Corporation is seven separate, interrelated businesses. Sound system installations have become a main line in the past decade. We've learned a lot. We'd like to share it in hopes it will make your job easier and your sound system better. We'll simplify with a "grocery list" of do's and don't's.

DO ask pointed questions about recent installations and get references. Find out about competence, professionalism, and delivery. Beware of reports of equipment delivery delays. Such delays are usually a result of credit problems. Almost all equipment can be delivered immediately if the books are clean.

DO look at your prospective installer's shop. It should have test equipment including: Distortion Analyzer' Impedance Bridge; Oscilloscope; Precision Sound Level Meter (with Octave Band Filters); several Multimeters, preferably including a VTVM; and Spectrum or Real Time Analyzer (1/3rd octave or better). Installers without this equipment are probably not committed enough to handle a state-of-the-art system properly. Look for a professional drafting table and someone who knows how to use it.

DO ask to see previous installations. Look for neatness and order. Most importantly, LISTEN.

DO buy locally. This ensures quick service on equipment. Installers who are charged with installing professional gear, bought out-of-town at a lower price, may not be too available when it breaks down. Try to do business with an installer who sells only the top lines, can deliver them now, and will service them later.

DO beware of sideline installers . . . recording studios who sell equipment as dealers in order to get equipment cheap to put in their studios. This is not to say that all such studios should be avoided. Many professional studios, such as Alpha, are ideal because they have millions of dollars worth of equipment under one roof for testing and proving, and can attract the brand of talent necessary for top notch installations. Your best line of defense here is to ask questions and probe.

DO inquire as to the talent who will melt the solder. Are the actual firing line laborers part-timers? Ex-roadies? Temporaries? Make sure your wirers are pros.

DO require that the final system tuning is done by qualified personnel. Require that such personnel have successfully graduated from a course on sound systems and equalization, such as those given by Altec, Dukane, JBL, Rauland, Syn-Aud-Con, etc.

DON'T necessarily take the lowest bid. The equipment all costs pretty much the same. Low bidders cut back by hiring cheap labor. Cheap labor does cheap work. Eventually the lowest bid will not prove the wisest move. Read all the DO list above before considering the lowest bid.

DON'T do business with a buddy **because** he's a buddy, or because he **needs** the business. You may have to sue your buddy!

DON'T overburden an installer with a job the installer can't handle. It will probably destroy his business and your sound system.

DON'T buy more or less system than you need. Go with the professional firm that asks the most intelligent questions about your needs, present **and** future. You'll be glad you did.

Finally, write a tight spec! Require things like system documentation, proper installation techniques, test requirements, coverage and frequency response requirements, neat labeling, perhaps a bond or even cash deposit. Buy and read a copy of Don & Carolyn Davis's "Sound System Engineering." Write your specs according to the final chapter of that book and you can't do wrong. If your installer doesn't know the book, you can't go right. It's one important standard to guide you.

If you need further information, call Alpha Recording Corporation.

Dave Jones (L) with Bill Sprinkle (C) and Joe Horner (R). Bill Sprinkle has all the fine attributes of his famous father, Mel Sprinkle. ♦

TEF® TALK

The latest news from Dick Heyser is that he now has a disk programmed to give 2.5 millisecond *full scale* ETCs that match those you see in his reviews in *Audio* magazine. Dick reports that the TEF® analyzer is allowing him to program tests that in the past could only exist in his consciousness. We suspect that the names Heyser and Stanley will one day be as universally recognized as Fletcher-Munson and Steinberg-Snow, of the original Bell Laboratory fame. ♦

SYSTEM IMPEDANCE DEFINITIONS

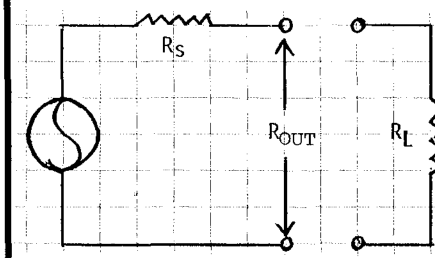
Electronic components, passive devices and various electroacoustic transducers all have both "rated" and "actual" impedance values. In evaluating impedances, let's work from the input to the output of a typical sound system.

Microphones

Early Western Electric microphones contained transformers that required careful termination at the input to the mixer amplifier. Contemporary microphones prefer working into what amounts to an open circuit, i.e., some value at least 10 times their rated Z.

A representative moving coil dynamic microphone looks like Fig. 1. Microphone available input power figures are calculated from the "open circuit" voltage sensitivity and the *rated* or *specified* impedance (R_{OUT}).

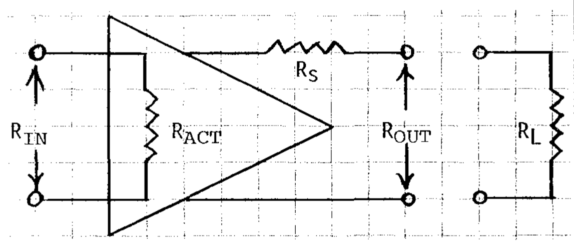
FIG. 1 MICROPHONES



TYPICAL VALUES

SOURCE IMPEDANCE	$R_S \approx 110\Omega$
SPECIFIED OUTPUT	$R_{OUT} = 150\Omega$
PREFERRED LOAD	$R_L \geq 1500\Omega$

FIG. 2 MIXER AMPLIFIERS



TYPICAL VALUES

SPECIFIED INPUT	$R_{IN} = 600\Omega$
ACTUAL INPUT	$R_{ACT} = 3000\Omega$
SOURCE Z	$(R_S) = 110\Omega$
SPECIFIED OUTPUT	$R_{OUT} = 600\Omega$
PREFERRED LOAD	$R_L \geq 600\Omega$

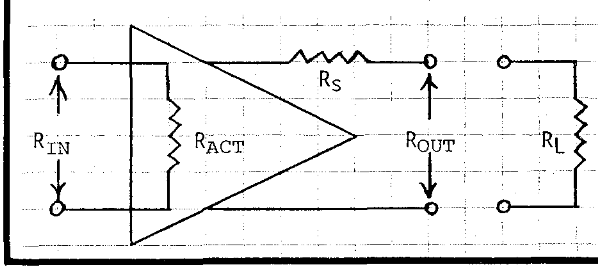
Mixer Amplifiers

Mixer amplifiers both mix and amplify, usually to line level but occasionally to power level. Microphone inputs on mixers are normally labeled between 150Ω and 600Ω, R_{IN}. The R_{ACT} is usually equal to or greater than 10 times the specified value. This is an intended situation and modern microphones work best under these impedance conditions. (See Fig. 2)

Mixer output impedances are commonly rated at 600Ω, R_{OUT}. Their actual source impedance, R_S, is usually lower (on the order of 90 to 130Ω). The mixer will deliver its indicated available input power when connected to an R_L equal to or greater than its R_{OUT}. In addition to its source impedance, R_S, it's necessary to know if the output is balanced or unbalanced should it become necessary to "build out" R_S to a higher value. In the unbalanced case, the build out resistor, R_b, is equal to desired $R_D - R_S$ and goes in the ungrounded side of the signal path. In the balanced case, there are two R_b's which are equal to $0.5\ (R_{desired} - R_S)$ and one goes in each side of the signal path.

Line Level Amplifiers

FIG. 3 LINE LEVEL AMPLIFIERS



TYPICAL VALUES

SPECIFIED INPUT	$R_{IN} = 600\Omega$
ACTUAL INPUT	$R_{ACT} = 2800\Omega$
SOURCE IMPEDANCE	$R_S = 110\Omega$
SPECIFIED OUTPUT	$R_{OUT} = 600\Omega$
PREFERRED LOAD	$R_L = 600\Omega$

These devices may come along for the ride in active equalizers or they may be separate "gain" packages intended to follow a series of "loss" circuits. Sometimes they are used as "unity gain" devices that provide isolation between circuits. Typical devices have inputs specified as either 600 or 15,000Ω. The 600Ω rating applies when a "matching" transformer is plugged into the line just ahead of the input circuits. Line level amplifier outputs normally reflect mixer amplifier practices being specified 600Ω out, i.e., with an internal source impedance of approximately 90 to 130Ω. (See Fig. 3)

Power Level Amplifiers

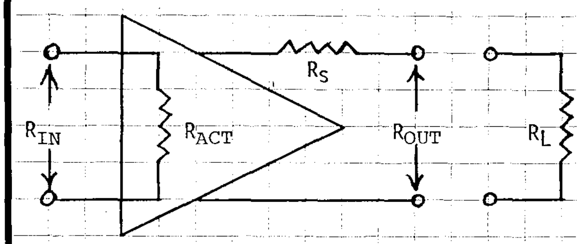
The input circuits of power level amplifiers tend to follow the same pattern as the input of line level amplifiers, i.e., 600Ω or 15,000Ω, depending on the presence or absence of a "matching" transformer. Today the transformers may be replaced by "plug in" active devices that provide the desired impedance and level adjustments.

Continued next page.....

It's at the output of the power level amplifiers that the bill comes due. Up to this point, "voltage swing" has been the main concern with little thought about power. At the interface between each component, very little current flow has been present, i.e., at the inputs and outputs of the individual devices. But, as we have stated earlier, the entire idea of a sound *system* is to raise and distribute acoustic power, not voltage swings.

Typical good practice at the output of power level amplifiers is to provide a low source impedance $R_S \leqq 0.1\Omega$.

FIG. 4 POWER LEVEL AMPLIFIERS



TYPICAL VALUES

SPECIFIED INPUT	$R_{IN} = 600\Omega$ or $15{,}000\Omega$
ACTUAL INPUT	$R_{ACT} = 2800\Omega$ or $15{,}000\Omega$
SOURCE IMPEDANCE	$R_S \leqq 0.1\Omega$
SPECIFIED OUTPUT	$R_{OUT} = 8.0\Omega$
PREFERRED LOAD	$R_L \geqq 8.0\Omega$

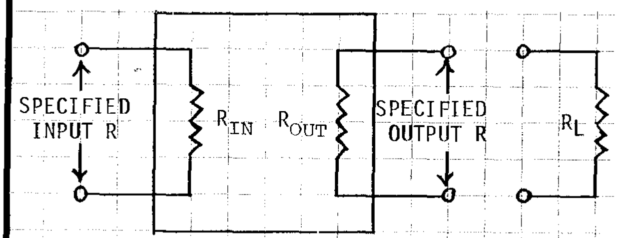
The specified output impedance will typically be 4, 8, 16Ω along with 25, 70 and, in Europe, 100 volt outputs. On occasion (Indianapolis Motor Speedway and the now past history Ontario Motor Speedway), 200 volt distribution has been used to keep current amplitudes in check.

Since $R = \frac{E^2}{W}$ the minimum R_L for the constant voltage systems is dependent upon the power rating, W, of the amplifier with E being the output voltage rating. (See Fig. 4)

Passive Networks

Typical passive networks include conventional crossover networks between the power level amplifier and the loudspeakers, a low level passive crossover installed just ahead of the two or more power level amplifiers used, a passive equalizer, attenuators and pads, and signal distributing networks (splitters).

FIG. 5 PASSIVE NETWORKS IN AUDIO "LINK" CIRCUIT



TYPICAL VALUES

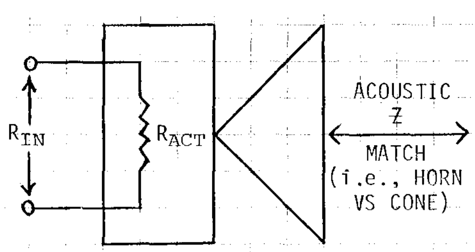
SPECIFIED INPUT	$R = 600\Omega$
ACTUAL	$R_{IN} = 600\Omega$
ACTUAL	$R_{OUT} = 600\Omega$
SPECIFIED OUTPUT	$R = 600\Omega$
REQUIRED	$R_L = 600\Omega$

These passive devices *require* exact impedance matches or they no longer perform to specifications, i.e., the crossover networks operate at new unwanted crossover frequencies, attenuators and pads become uncalibrated as to level, equalizers present bizarre new amplitude responses, and reflections of transients at the mistermination points degrades quality audibly. (See Fig. 5)

Loudspeakers

Both loudspeakers and microphones present a dual impedance problem: the acoustic impedance and the electrical impedance. Here we are treating the electrical impedance. In the case of loudspeakers, their impedance can vary appreciably with frequency and the normal specified Z is the lowest reading just above resonance for the device.

FIG. 6 LOUDSPEAKERS



TYPICAL VALUES

SPECIFIED INPUT $R_{IN} = 8.0\Omega$

ACTUAL INPUT R_{ACT} = VARIES WITH FREQUENCY - NORMALLY SHOULDN'T RUN BELOW R_{IN}

Good practice would dictate that the loudspeaker be connected to an output terminal on the power level amplifier that is equal to or below the loudspeaker's specified Z, i.e., R_{OUT} of amplifier $\leqq R_{IN}$ of loudspeaker. (See Fig. 6)

Past practices have included the series parallel connection of large groups of loudspeakers without realizing that such connections inadvertently modify the coverage angle of the devices involved.

Summary

Impedance selection and adjustment is not difficult and an orderly efficient set of choices distinguishes the professional sound system engineer. It cannot be emphasized too much that specified values are just that and

Continued next page.......

nothing more, and it is mandatory *that the actual values be measured.* Be sure to acquaint yourself with the description of "constant voltage" and "constant current" output circuits. The expectation that you are across one when actually it's the other leads to rapid confusion.

Finally, remember that the impedance is but one of the two factors requiring careful analysis at each interface point in a *system.* The other is level. Be sure that the practices being used in both cases are compatible with each other: an impedance change to adjust level that uncalibrates a passive attenuator in the same link circuit is an example of incompatible practices. ♦

CLAIR BROTHERS

Excerpt from "Pro Sound News" (June)

According to Roy Clair, two systems were designed to satisfy both sides of the horn-coloration issue. What is unique about the new systems is that the new acoustical measurement techniques, including the Tecron TEF Analyzer and computer program for maximization, were exployed in their design.

We were very pleased to see that the Clair Brothers of Lititz, PA. were designing their newest loudspeaker systems using the Tecron TEF® Analyzer. The Clair Brothers have always been noted for their ability to produce above average road show systems and the combination of their experience and TEF analysis should result in greatly improved show systems.

Also, the new enclosures use new materials and are sealed in back. According to Clair, they are quiet from behind. (Emphasis ours.)

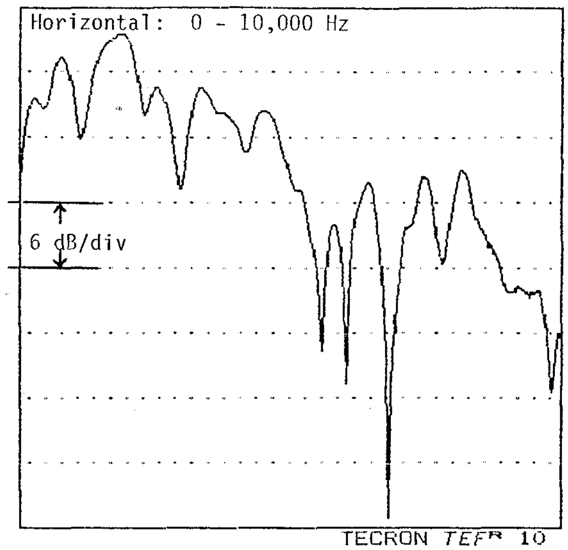
For those of you who have attended a Syn-Aud-Con class in the last year or so, does this ring a bell? ♦

PHASE CANCELLATIONS

This TEF® measurement was made at the mixing console in a large multipurpose auditorium. The loudspeaker system is composed of excellent equipment. BUT, it is a perfect example of what happens when a complex array is installed without a way to measure the interaction of the horn. Needless to say, the frequency response of the system is different in different areas of the auditorium.

The first balcony, for instance, has little other than highs.

The second balcony approaches the sound that the system should produce. ♦



OXYMORONS

"Oh, *nobody* goes there anymore; it's too crowded." is an oxymoron. "Oxy" means sharp and "moron" means stupid in the ancient Greek, so oxymoron is itself an oxymoron.

JOHN LANPHERE recently sent us a newspaper clipping on this interesting word. Paul Klipsch has collected oxymorons for many years and includes in his collection:

"Brass angle irons" .... "Military intelligence" .... "An officer and a gentleman"

and hundreds of other self-contradictory terms.

A new one to us is: "Federally funded projects on self-reliance"

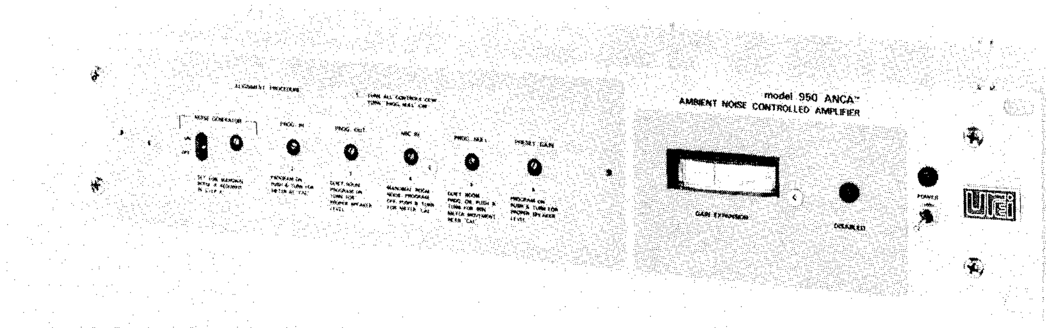
We'd be especially pleased to publish audio and acoustic oxymorons. The best place to initially search for them would be in the titles of AES papers:

"Voltage decibels" .... "Arbitrary dBm" .... "Digital audio"

are examples that come immediately to mind. ♦

AMBIENT NOISE CONTROLLED AMPLIFIER

ANCA (Ambient Noise Controlled Amplifier) allows automatic control of the *level* of a sound system controlled by the ambient noise level present in the environment where it is used.



The UREI Model 950 ANCA is unique in that it allows real time dynamic control by being able to tell its own sound system signal from the ambient noise level present (thanks to a clever "nulling" circuit). It also has a "lock up" circuit available at the throw of a switch that provides for dynamic adjustment of the system's level until the "page" actually begins and then "locks up" the level that was present just before the system was seized for the announcement.

The only caution necessary when using these devices is to be sure that the power available and the power handling capability of the transducers can handle the gain increases called for automatically after you leave the system set up. These devices work well at airports, industrial plants, and other sites where the ambient noise level varies widely and frequently. ♦

TEF® ANALYZERS:

AUDIO AND ACOUSTIC MEASUREMENTS

Having now spent the past five years using various TEF® measurements and particularly the application of this measurement technology to the analysis of the audibility of various acoustic parameters in spaces, we feel the necessity to stress the fundamental underlying philosophy behind all meaningful measurements.

FACT: Possession of or access to computers, programs, advanced analyzers, tools, etc., in other words, reams of new data, is worthless without the accompanying intellectual effort to understand the underlying principles behind the questions that preceded the attempt to "measure" an answer.

Watching the "myth" information of the intellectually inadequate users of the devices, and the worshipful acceptance of nonsense from the high priest printout, all leads to our desire to again point out the following honest waymarks for the sincere professional in search of true personal maturity in these technologies.

1. Devices have nothing to do with understanding.
2. Understanding comes from mentally coming into "resonance" with the thought that expresses a fundamental truth.
3. Reading and reading ability (translate as ability to comprehend the written word as a "mental image").
4. Those who seek "learning" by viewing rather than "chewing" will remain among the witnesses of the superficial.

What all this means is that we sincerely believe that the real value of TEF analysis lies in the study (in the *written* literature) that it forces on the individual attempting to use the process and that the conceptual breakthroughs that are occurring and that will occur with increasing frequency are the result of the "stirred up" of thinking about familiar problems, not just "viewing" them in the new forms of display.

A first peek at some of the subjects that should excite a beginning TEF user are:

1. The construction of the analytic signal. (The real and imaginary parts.)
2. The study of phase measurements, such as, Nyquist, Bode, and Nichol plots.
3. The partitioning of kinetic and potential energy in a dynamic system.
4. The fundamental constraints *on all measurement systems* with regard to low frequency resolution and ellipsoid size.
5. An absolutely clear mental image of the role of time windows with incremental time steps along with instrument bandwidths vs actual frequency resolutions.
6. A sufficient understanding of FFT hardware to understand why it cannot do what is required to obtain meaningful acoustic measurements easily done by a TEF analyzer. This is important because FFT analyzers display "something" when asked to and you must be able to discriminate between meaningless displays and those pregnant with meaning. (See Gerald Stanley's article on page 8). ♦

"THE NEXT DECADE IN PROFESSIONAL AUDIO"

We were rummaging through an old file recently and came across an article we called "The Next Decade in Professional Audio" -- meaning the 1980's. It was written in 1978 and published in 1979 by *Sound & Communications* with a new title, "Audio's Tomorrows." Maybe the publisher didn't think it would happen before 1990. What do you think?

AUDIO'S TOMORROWS

by Don Davis

Any industry belongs to its innovators. When an innovator is inventive and entrepreneurial as well, then he is called a "good businessman." Often innovators require association with a team that includes men possessing these other qualities, and because of their individualities such teams are quite often interesting seas of conflict united to achieve a goal at least partially visible to all the participants.

Such teams are in no way part of the concept of "professional management" (whatever that term might actually mean). The "bottom line" is really not part of the thinking of the innovative, inventive, and entrepreneurial personality. These men are pressured by ideas, not inventories; motivated by accomplishment, not applause; and are leaders of those stout enough to follow, not managers of opportunit existing alone in a highly paid post.

Audio has benefitted from many such men. Some have flowered in environments like the Bell Telephone Laboratories, Hewlett Packard, and other giants that on occasion provide haven for the creative individual; others have been the founders of companies or are talents that wandered throughout the industry. The audio industry is replete with their memories and memorials in successful companies, products, and people they have developed.

The presence of one or a group of these men is manifested by unique, useful, and usually successful product ideas that tend to lead the industry along new paths. On occasion, truly great ideas do not come to fruition at the time of their originator's conception, due to lack of adequate materials, necessary supporting software, or failure to find access to all three qualities: innovation, invention, and entrepreneurial capability. This is why many of the best "new ideas" can be found in the "old literature." The identification of a new idea in the minds of manufacturers (rarely the source of the idea, however) leads to active, feverish, commercial development of every conceivable variation of it to the ultimate benefit of those astute enough to select among the best offerings for use in developing better and better audio systems.

The computer has helped clarify the difference between a "component" (hardware) and its "application" (software). The audio systems engineer uses the hardware offered by manufacturers to develop software that efficiently solves the problems for the end user of the system. In the development of this software, the sound system engineer will, in turn, creatively generate *techniques* for design, installation, and maintenance of the system as a useful tool for the end user.

In this article I would like to suggest ten areas that, during the next ten years, promise such activity in terms of hardware (components) and software (techniques) likely to dramatically change professional audio.

1. The Directivity Control of the Loudspeaker

The best of the modern loudspeaker (designed in the 1920s by Wente and Thuras) has undergone steady evolution, but not revolution. Its greatest flaw in the 1920s was its inability to reproduce the directivity characteristics versus frequency of the instrument recorded. Fifty years later, literally zero progress has been made in achieving this goal (which, incidentally, was originally stated as a goal by Wente and Thuras).

Amplitude vs frequency, distortion vs frequency, power handling and reliability have all been improved over these fifty years but control of the polar response is still a whim of the geometry of the various devices used to couple the transducer to the air.

Digital techniques will make it possible to build a loudspeaker array capable of reproducing dynamically the identical directivity pattern, frequency by frequency, of any sound source.

The accomplishment of this goal would finally achieve that long-sought-for result of a talker sounding like a talker when reproduced over a loudspeaker in a small room.

2. Fully Digital Recording

The industry is at the present time "playing" with this idea. Various digital processing techniques are employed using the tape as the storage medium.

What is truly required is a computer with sufficient memory to "store" the entire piece of music in its own non-volatile, all-electronic registers. Flutter and wow, signal-to-noise, distortion, etc., become meaningless, unmeasurable.

3. Holomorphic Records

While the technique described above could be transferred over various high quality link circuits to other memories (dial-a-symphony, where the master computer "dumps" the desired recording into your terminal's memory) there will still be those who desire a convenient method of storage (for archival purposes, for example) and/or transportation to a computer not accessible over high quality link circuits. The Holomorph recording is the answer.

"When we take a hard look at our present discs and tapes, we find that they are not very efficient in their use of the medium on which they are placed. On a disc, for example, the entire audio program is on that long filamentary chip which cutting engineers vacuum up and throw away. We are left with a cast of the now departed audio program." (Dick Heyser)

The holomorphy recording, which could be a piece of film-like substance, would not be time-serial like records and tape but every portion of it would contain the *entire* recording. This "record" when scanned with a suitable scanner would once again fill the memory registers of a computer with the desired program material, which could then be called out for time-serial listening.

4. Multi-Mega-Mega Byte Battery Operated Hand-held Computers

With the current development of electronic storage registers approaching the density of the human brain's memory storage, it is not unreasonable to expect that attache-case size, battery-operated, alphanumeric, IBM 360 or 370 capacity, hand-held computers will become available at prices under $10,000 (today's dollars, that is). Access to such capacity in a convenient form will lead quickly to the reform of mathematics as we now know it.

If you were at a party and a new book was being described to the group and you were to remark, "I can't read," what would be the verdict of a majority of your listeners? Yet, if we were at the same party and a new dirac delta were described and you stated, "I've never been able to understand math," a majority would immediately echo your statement. Mathematical illiteracy will no longer be overlooked as normal in the near future. As mathematical literacy spreads, due to the advent of the low-cost, high capacity computer, it will cause an even more pronounced change in human thought than the spread of literacy did with the advent of the printing press. Rough times are ahead for "mental stick-in-the-muds."

5. Cybernetic Transduction

The actual interconnection directly to the human brain may be more than ten years in the future, but this writer would not be astounded to see it happen near the end of the coming decade. The digital computer will directly feed the signal to the brain in the same manner as our aural nervous system does now.

Imagine, if you can, the instant A-B between the recorded mental signals of a clear brain and an intoxicated or drugged brain, or that of a genius or artist thinking through his problems. Distortion in such signals could take on new significance as could the search for privacy.

6. Two Channel Surround Sound

This has already been demonstrated by Manfred R. Schroeder, when he developed a "kind of super stereo." He has written, "Because we have only two ears (and since head motion is not crucial in localization), two loudspeakers should suffice to evoke all the proper perceptions of acoustic space—provided the sound waves are "tailored" in such a way as to produce, at the listener's eardrums, pressure waves indistinguishable from those that the ears would have received in a free sound field, and to the extreme sides." (IEEE)

Schroeder then proceeded to solve the riddle of the crosstalk caused from ear to ear by the diffraction of sound around the head, discovering in the course of the solution that the crosstalk paths lay several layers deep. His comments relative to the results were: "Practical experience with the filtering scheme ... has been nothing less than amazing. Although the loudspeakers are the only sound sources (two loudspeakers 30° apart), virtual sound images can be created far off to the sides and even behind the listener. In fact, even the elevation angle of a sound source is properly perceived (by people with proper head shapes)."

These experiments reveal the true poverty of thought involved in Quad.

7. Truly Active Acoustics

Imagine a wall electronically adjustable for any desired amount of absorption. Simply dial in the desired amount of absorption and adjust the surface area until a recording studio control room meets the desired acoustic criteria. Microprocessors, sensing the impinging energy, control a series of wall transducers that provide either nulling or reinforcement of the energy detected. Today such systems as "assisted resonance" are working models of the potential that could be realized in the future.

8. Automatic Digital Equalization

Again, the development of very low cost omnipresent microprocessors will lead to loudspeaker-room equalization that continuously compares the input to the system and adjusts the output so that the signal measured at points at the audience area receives the desired variation, if any, at the listener's ears. Automatic equalization is held back only

Continued next page.....

by the fact that a majority of sound systems still are so poorly designed as to be unable to be used with equalization without problems being magnified.

9. Ultimate Anti-Feedback

The ultimate answer to the acoustic feedback problem is to find a way to turn a "closed loop" into an open loop, so far as feedback is concerned. Suppose that as your voice traveled to the microphone and on into the sound system, a code was impressed on the signal that would serve as an identification that it had passed through the system. Then when, after multitudinous reflections, it arrived back at the microphone and attempted to re-enter the system, a processing device separated those signals without encoding and sent them through the system and rejected all coded signals. In other words, a closed loop for new information but an open loop for any redundant information—a very fast real time autocorrelator.

10. TDS-PRP-FFT as Related to Recording Control Rooms, etc.

Microphones are about to be *properly* used in musical recording for the first time since their invention. The basic error here has been the profuse development of every conceivable type of microphone for the recording of music in only the free field or random incidence form. By this I mean that the microphone's response has been adjusted to optimum performance for use in either a free field or a semi-reverberant sound field. More than likely this occurred through the philosophy that the microphone was a sort of "model of the ear," forgetting the marvellous computer that comes attached to most ears. Fortunately—or there would have never been a recording industry—most early microphone use kept the units relatively remote from the nearest reflecting surface (a standard microphone stand brings the unit to a height of about five feet), thus placing the effects of such surfaces high enough in frequency to be of minor importance.

Another tendency, early acquired, was to over "deaden" the reflecting surfaces and thus use the microphone in an *actual* free field, in spite of the obvious dislike our ears have for such "dry" environments and the ear's preference for "warm" reverberant spaces when listening to "live" music. The answer to this dilemma has just been discovered and is called the "pressure response pickup" technique (PRP). (Process developed by Ed Long and Ron Wickersham.)

In essence, it takes microphones designed for optimum response in a pressure environment and then proceeds to record with them in the pressure environment at the very surface of any large boundary. The miracle achieved is full preservation of the amplitude and the time-phase information in the room minus the reflected anomalies between the microphone and the boundaries, but with reflected sound, direct sound mixtures of the sound source.

This new technique is literally in its infancy. Imagine what fantastic opportunities lie just over the horizon for those able to meaningfully adapt this technique to the demanding requirements of real life recording in all of its manifestations.

Conclusion

These short discussions of ideas already large in the womb of current technology by no means pretend to completeness or comprehensiveness. They are simply ten thoughts out of the infinite store awaiting recognition.

In over 100 Syn-Aud-Con classes, we have had the opportunity to serve as the "toolmaker" for literally thousands of audio engineers eager to participate in the future and aware that they must know the best of the past and present in order to contribute meaningfully to tomorrow.

Through being introduced to time delay spectrometry, real time analyzers, modern scientific array design, equalization, acoustic room design, etc., they become familiar with the basic tools as well as the more sophisticated ones. Perhaps most important of all, they open an avenue in their lives for participation in their industry unique in the history of audio.

Mr. Davis heads Synergetic Audio Concepts, a consultant firm which offers professional seminars. ♦

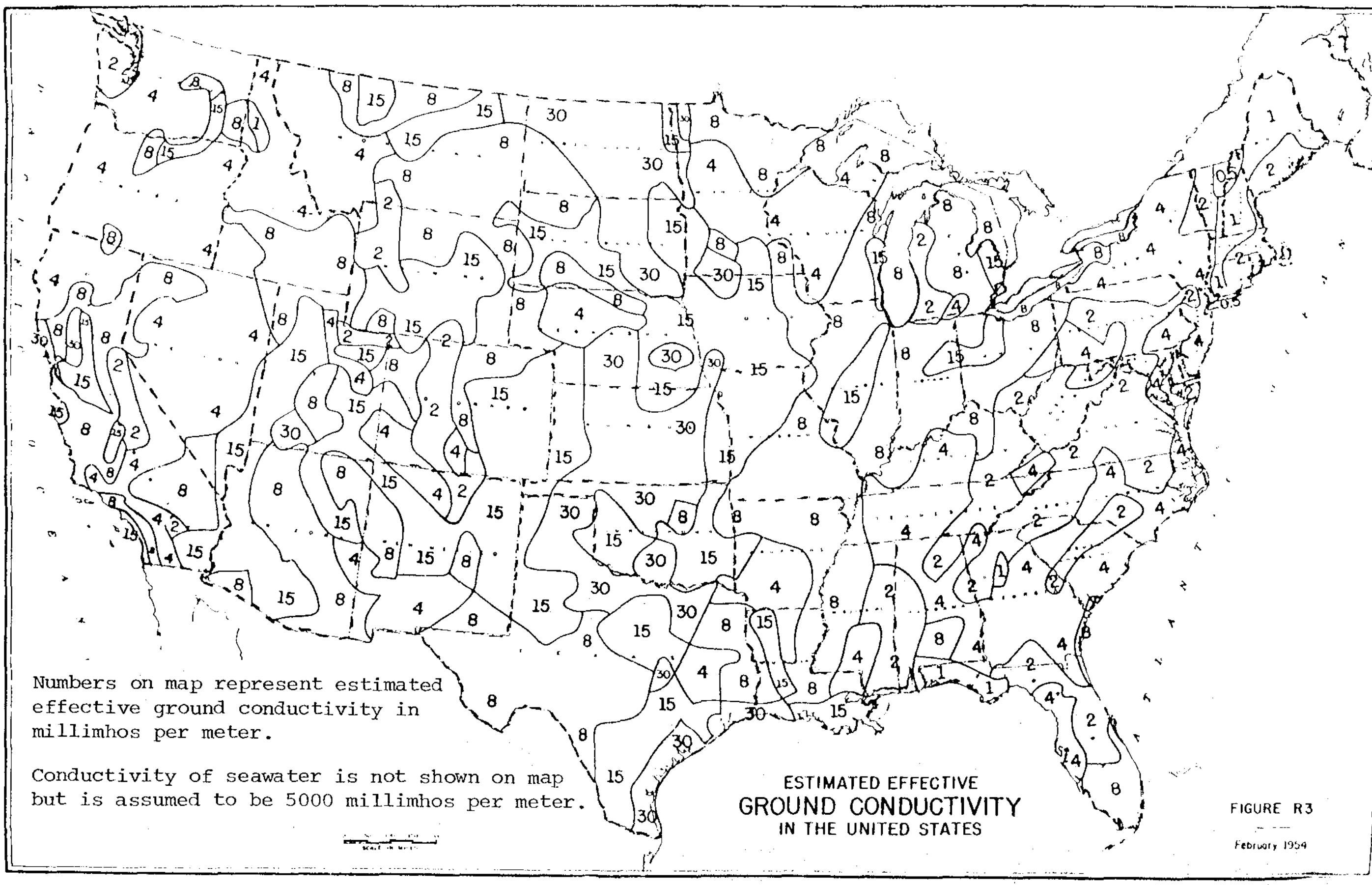
THE MATCHBOX

We are always pleased when we encounter a new and useful form of "black box." We are reproducing the data sheet received on this latest unit. We have not run tests on this unit yet, but now have one in our hands and will do so. The unit we received seems well constructed and the spec sheet is written and correctly uses the terms presented. Because this unit fills such an obvious gap in the field of commonly interfaced apparatus, we suggest that a test of one of these relatively inexpensive units ($195.00) is a worthwhile experiment. ♦

GROUND CONDUCTIVITY IN THE U.S.

The map shown here is of the "Estimated Effective Ground Conductivity in the United States." Conductivity of seawater is assumed to be 5000 millimhos per meter. The mho is the unit of conductivity and is the reciprocal of the ohm. Thus, a conductivity 5000 millimhos per meter is the same as a 1/5 = .20 ohms per meter.

Along the Southern California seacoast, we have 15 millimhos per meter or 1/.015 = 66.67 ohms per meter. ♦



MICROWAVES CAN DAMAGE CHROMOSOMES

The following excerpts were taken from *The Institute*, Volume 7, Number 8, News Supplement to IEEE Spectrum:

Boulder, Colo.--Microwave radiation at power levels well below those permitted by current American safety standards can produce chromosomal damage in the sperm cells of mice, according to recent experiments performed at the U.S. Public Health Service's National Center for Devices and Radiological Health in Rockville, Md. A different but related set of experiments showed that radiation at power levels only 2.5 times higher than current standards could more than quadruple spontaneous abortion rates in exposed mice.

New findings on the possible hazards of strong 60-hertz power-line fields were also reported at the BEMS meeting. Experiments showed that fields at strengths comparable to those found near high-voltage power lines have induced brain abnormalities in rabbits exposed prior to and immediately after birth and have caused a doubling of abnormal fetuses in miniature swine after many months of continuous exposure. (BEMS is the abbreviation for Bioelectromagnetics Society.)

Among the results indicating possible hazards of power-line fields, perhaps the most dramatic were those presented by Richard Phillips of Battelle Pacific Northwest Laboratories, Richland, Wash. Dr. Phillips reported on a recently completed four-year-long experiment in which Hanford miniature swine were bred in 60-Hz fields of 30 kilovolts per meter. The field strength was selected to produce currents in the pigs like those produced in humans by 10-kv/m fields, comparable to those directly beneath high-voltage (for example, 765-kv) transmission lines. After 18 months of exposure, swine were found to have twice as high a rate of abnormalities in their offspring as did the controls.

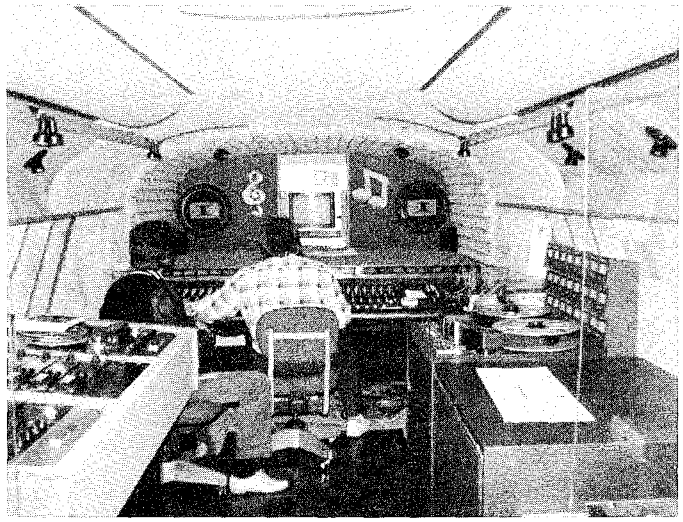
While these new findings on interaction of electromagnetic fields with life processes will have to be more fully explored in future experiments, many participants at the BEMS conference thought that the steady accumulation of evidence will force tougher safety standards for both radio-frequency and power-line-frequency fields.

Eric Lerner ♦

"COVER YOUR FACE"

Neil Muncy says there is an audible improvement from covering the face of the loudspeaker monitors. Even the port is covered on the UREI 811.

Note the use of Polycylinders on the ceiling and "Haas Kickers" on each side of the bus at the left and right corners (especially visible right) of the picture.



Skyelabs' bus, The Rover, built by Bob Skye (Dover, MD.) with the help of acoustician Neil Muncy. ♦

BOOKS OF INTEREST

"The Complete Guide To Satellite TV"

This is a worthwhile survey or overview book. Satellite "footprints" are much like a loudspeaker coverage contour plot. We found this book to be useful in a tutorial sense, thanks to a collection of mathematical design equations. The author, Martin Clifford, has an excellent grasp of what the average calculator owner is ready for in this type of book.

Satellites will play an increasingly important role in our lives, though not in television in our humble opinion. It's in telecommunications - teleconferencing that we foresee significant changes in our audio industry. This book represents a good basic primer of what kind of engineering is required. TAB BOOKS, Inc., Order No. 1685, $17.95 hardbound; $10.95 paperback. ♦

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