SYNERGETIC AUDIO CONCEPTS

newsletter

P.O. BOX 427, TUSTIN, CALIFORNIA 92680

VOLUME 7, NUMBER 1
FALL, 1979

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

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SYN-CON-CON NEWSLETTER. Published quarterly by Synergetic Audio Concepts, Don and Carolyn Davis, Editors, P. O. Box 1134, Tustin, CA 92680. Application to mail at Second class postage rate pending at Tustin, CA. Subscription price, on year, to United States and possessions, $25.00; Canada, $30.00; all other countries, $31.00. Not sold by single copy, available only by subscription to graduates of Syn-Aud-Con seminars.

1979-1980 SEMINAR SCHEDULE

Future Syn-Aud-Con seminars will be held in the Southern California area. Currently we are holding our classes in the beautiful Dana Point Harbor area, at the Dana Point Marina Inn (near San Juan Capistrano).

We have added an additional day for those who would like to come in one day in advance to "learn the basics". The day of basics will start in January and will be conducted by Farrel Becker and Ken Wahrenbrock, assisted by Don & Carolyn Davis and Gina Becker. The regular 3-day seminar will be conducted by Don Davis, assisted by Ken Wahrenbrock, Farrel and Gina Becker and Carolyn Davis. You may attend either the three or four day seminar.

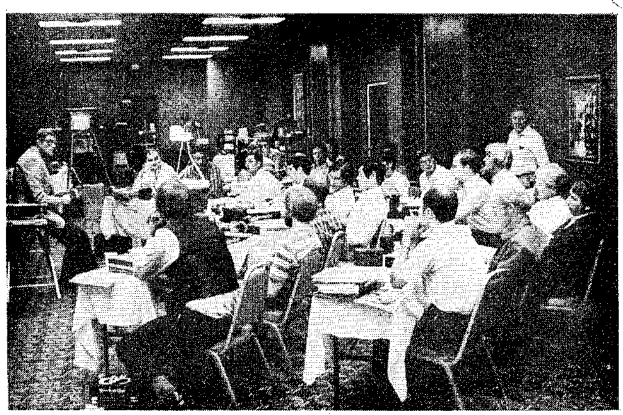
November 13-15, 1979	3-day Seminar	$500.00
January 22, 1980	Day of Basics	$125.00
January 23-25	3-day Seminar	$500.00
February 19, 1980	Day of Basics	
February 20-22	3-day Seminar	
March 11, 1980	Day of Basics	
March 12-14	3-day Seminar	
April 15, 1980	Day of Basics	
April 16-18	3-day Seminar	
May 13, 1980	Day of Basics	
May 14-16	3-day Seminar	
June 17, 1980	Day of Basics	
June 18-20	3-day seminar	

There may be some changes in the above schedule and format as we progress in our new mode of operation, but we will keep you informed.

ABOUT OUR NEW CLASSES IN S.CA.

As we enter our eighth year of service to the audio industry we are grateful to still be the leaders in our field. Our new classes, held here in Southern California right on the Pacific Ocean, now offer more advantages:

1. They are limited in size. Maximum attendance is in the low to mid-twenties.
2. Three instructors. We now have a greater diversity of teaching talent plus this allows you more face-to-face time with each instructor.
3. We supply lunches, dinners and morning and afternoon coffee breaks, as well as textbook, laboratory manual, and slide rules.



The attendance fee is $500 per person attending; $450 for Syn-Aud-Con graduates, and $25 off for 2 or more attending from the same organization. Add $125 to the above if attending for 4 days. The $50 allowance is made to Syn-Aud-Con graduates because we don't need to supply the textbook and lab manual. We are not able to continue our policy of half-price for graduates because of the smaller classes and the additional instructors for the class.

The classes will be from 8:30 a.m. to 5:30 p.m. each day with the evenings reserved for dinner, "hands on" experience, if desired, personal tutoring, or sharing of experiences with the other participants. The format of these classes is lecture-demonstration during the day with "hands on" opportunities in the evening sessions.

We use the Dana Point Marina Inn in Dana Point, CA right on the Pacific Ocean, next to historic San Juan Capistrano. Single rooms are in the $30 range. We will assist in arranging transportation to and from the airport, should you fly in.

Syn-Aud-Con and its graduates have literally pioneered 1/3-octave real time analyzers, sound system-room equalization, Live end-dead end (LEDE) control rooms, PZM™ microphony, and the licensing of Time Energy Frequency (TEF™) measurements (under the patents held by Dick Heyser and Cal Tech).

We are carrying on this leadership by having in our classes analyzers available only to TEF™ licensees: Analyzers that allow *you* to measure the direct sound and each reflection as to total spectral energy density, direction of arrival, time of arrival, and distance traveled -- all in essentially real time.

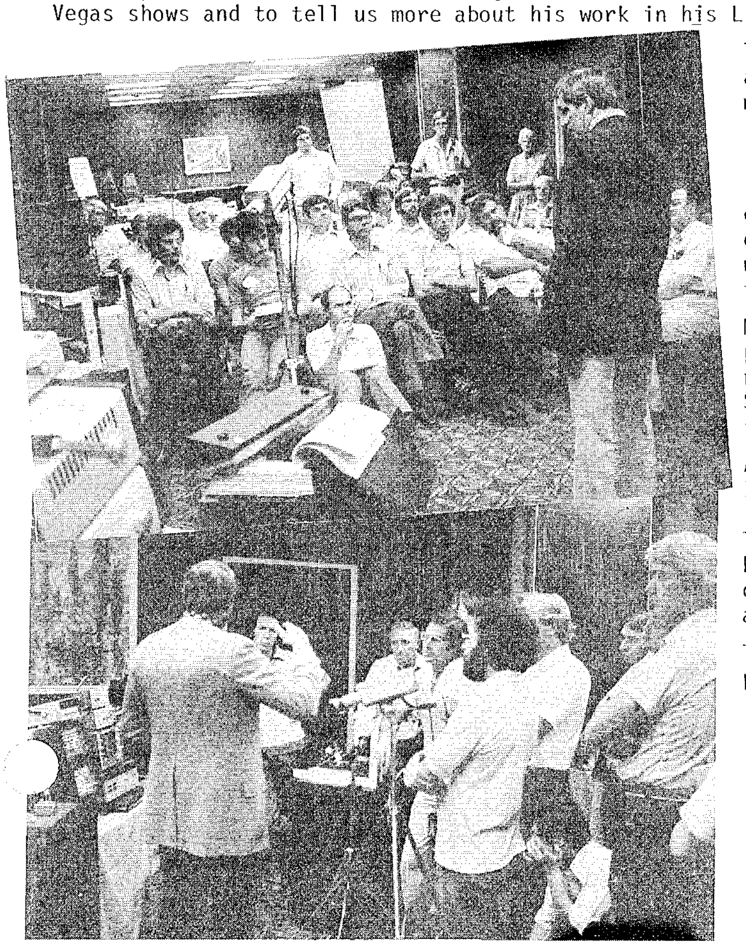
The classes are advanced in terms of new tools such as the latest HP 41C calculators specially programmed for these classes, TEF™ analyzers, and a host of "black boxes" designed by Ken Wahrenbrock of PZM™ fame and Farrel Becker, head sound man at Wolf Trap for the past three years, to make measurements rapid and accurate: the black boxes are an educational experience in themselves.

We'd be very pleased to have you join us.

THE SEPTEMBER CLASS

The September 1979 class in Anaheim, CA was put together by people calling in to ask why we were not traveling this Fall - and where could they attend a Syn-Aud-Con class. We told everyone if 21 people wanted to attend such a class we would have one in September in Anaheim. Within a short time 23 people signed for the class. Because we wanted to try several new things, we added an extra day to include everything - a field trip to an AES section meeting or dinner and a trip to Warner Brothers studios to h[illegible] see the new 3M digital machines; Richard Heyser dropped by the second evening for dinner and a couple of hours in which he outlined to the class Heyser's energy theorem (Heyser's fourth law of thermodynamics). (Heyser's unexpected visit was especially appreciated by the class.)

On the third day the class took a field trip to an especially beautiful new studio facility to make TEF™ measurements. And Chips Davis flew in from Las Vegas to share with the class his experiences in live sound reinforcement for Las Vegas shows and to tell us more about his work in his LEDE control room at Las Vegas Recording.



The class participated in a great deal of experimental approaches to studying audio and as a result were instrumental in helping us further strengthen our new seminar format.

First of all, limited attendance (no more than 24) is a very successful idea. We really get to know you and you get to know us. Just as successful is the 8:30 a.m. to 5:30 p.m. class hours. We discovered that the time for "hands on" experience is in the evenings. Participants want to see us use the equipment in well structured demonstrations before they attempt the same measurements on their own.

Many class members both new to the industry and experienced persons expressed the desire for a basic session ahead of the regular class, with an optional charge that would allow a Syn-Aud-Con many-time graduate to come in on the second day for the newer information.

An extra day will be added, starting with our January class. It will offer Farrel Becker and Ken Wahrenbrock as instructors with Don and Carolyn Davis and Gina Becker assisting. This extra first day is $125 additional and will only be held if a minimum of 10 sign up for it in advance of the class. (See Page 2 of the Newsletter for a full schedule and rates.)

This "Basics Day" will cover in detail with intensive personal tutoring:

1. How to use an electronic scientific calculator correctly
2. Ohms law, ac and dc
3. The decibel
4. Impedance
5. Basic audio terminology such as: RT_{60}, $S\bar{a}$, V, S, $\%Al_{cons}$, L_D, L_R, L_N, dBm, dBV, dB-SPL, dB-PWL, Wave analyzer, real time analyzer, FFT analyzer, phase and polarity, basic direction control devices such as columns, horns, cones, etc.

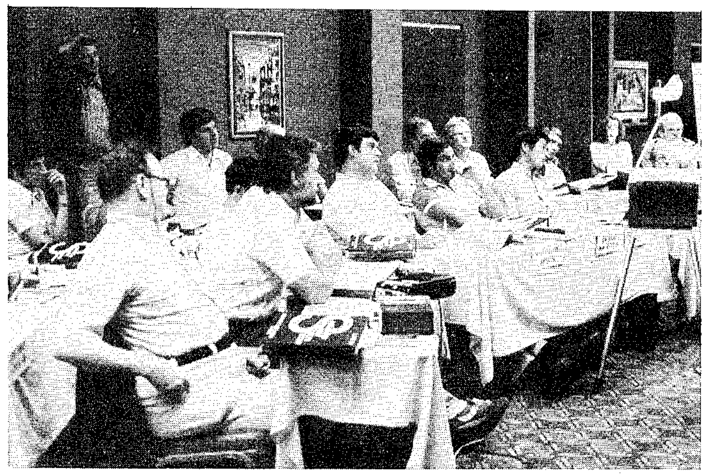
continued next page

September Class, continued

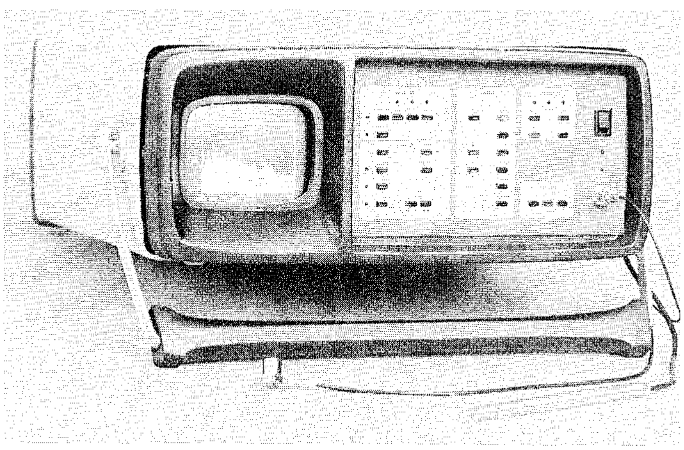
The next three days will assume that all members of the class are already familiar with these tools and will plunge right into their use in the design, testing and adjustment of all types of sound systems.

Any audio professional that has not yet been exposed to the latest TEF™ measurements is in for quite a treat. This technique is so superior to anything in your previous experience as to literally stagger the imagination. (And the latest TEF™ Workshop that Heyser conducted has opened avenues that we will be exploring for years.)

We used two classroom seating arrangements for the September class. One a semi-circle for lecture-discussion and at the end of the room we set up all the test equipment with plenty of open area to work around the equipment for demonstrations and hands on activity. This allowed the class to move about more, which was more relaxing and more informal.



REPORT ON THE GENRAD 2512 FFT



Syn-Aud-Con has chosen to use the GenRad 2512 FFT spectrum analyzer in our TEF™ measurement work. The many ETC photographs appearing in our current publications are from the screen of the 2512.

When we buy such a unit we, like you, have several concerns: the initial price, reliability, accuracy, ease of use, response of customers to the display format, back up by the manufacturer, availability in the field from rental agencies.

The GenRad unit has only one competitor lower in price, and we have not found any unit more reliable.

Accuracy seems to be a constant in all of the available machines. The GenRad unit is extremely easy to use, thanks to its displayed "menu" on the screen, chosen by moveable cursors. It is easy for TEF™ class members and consulting clients to relate to the display almost immediately.

All manufacturers seem about the same on service back up, application materials, etc.

We have established that the GenRad 2512 is available in any major city in the U.S. from rental agencies. Rental is about $1150/month. We just heard from GLEN MEEKS in Indiana (May TDS Workshop) that he has three studios to measure; therefore, he can justify the rental of the equipment for his clients. Dan Zellman, Howard Schwartz Recording in New York, has rented the FFT and associated equipment for measurements in two new studios and control rooms they are just finishing. GEORGE HORN, chief engineer of the new Criteria West studios, says that he will have a complete measuring chain available as they complete the construction of their new west coast studios. (Both Criteria and Howard Schwartz are TEF™ licensees.)

We are well pleased with our GenRad 2512 and have found it to be one of our most dependable instruments. We would have preferred 800 lines and two channels for the same money and I'm sure in three years we'll see such a creature. And there is a dedicated instrument for TEF™ measurements coming in the next year or so. In the meantime, we are acquiring new acoustical data daily and that's the real game.

PZM BULLETIN

We think you will like the PZM™ BULLETIN as it has a lot of material to help you explain the concept of the Pressure Zone Microphone to your customers. Note the pricing. KEN WAHRENBROCK is not establishing dealers in a legal sense, but he is offering protection to those Syn-Aud-Con graduates who are interested in resale of the Pressure Zone Microphones. Syn-Aud-Con graduates can purchase from Ken Wahrenbrock at the established net price, currently $320/pair for the PZM-130 models and $350/pair for the PZM-150.

In the next Newsletter we hope to announce a manufacturer of professional audio equipment, a Syn-Aud-Con sponsor, who will manufacture the PZM™ and distribute through their network of dealers. Until the new manufacturer is in full production, Wahrenbrock Associates will supply the PZM™. Even after the new manufacturer is in full production, Ken will continue to assemble and sell limited production models, as well as experimental models.

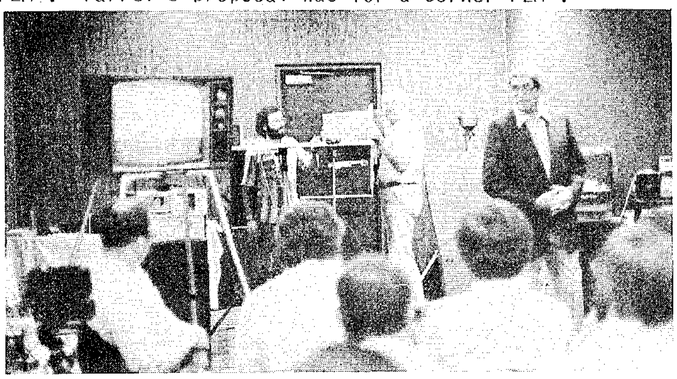
There is not a Syn-Aud-Con class but what some new use of the PZM™ is revealed. That is the excitement of Syn-Aud-Con -- exposing a new concept to the finest minds in audio. The feedback is explosive.

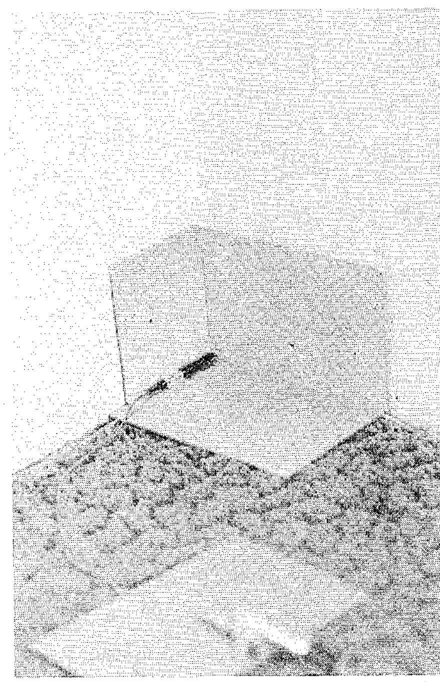
PZM ^{3}

OR

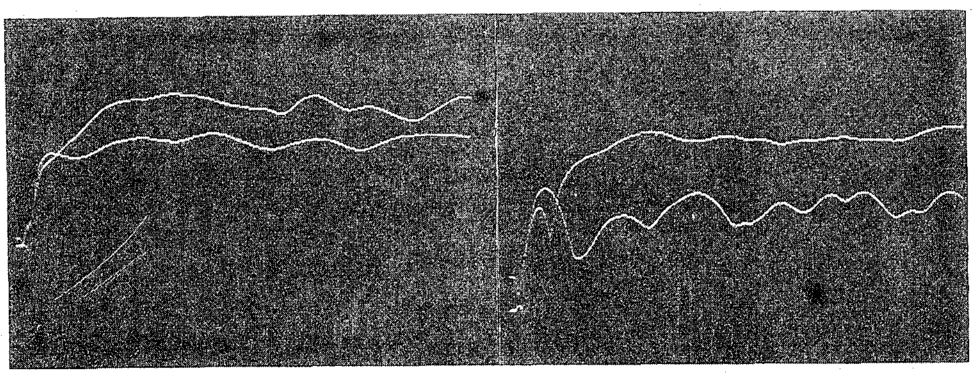
THE TRIHEDRAL PZM™ or K-PZM™?

In the Anaheim class in September, FARREL BECKER, working with KEN WAHRENBROCK, proposed a new version of the PZM™. Farrel's proposal was for a corner PZM™.





Farrel's reasoning was that "the PZM is not affected by the reflections from the surface it is mounted on. Therefore, if it is essentially mounted against *three* surfaces, as in a trihedral corner, it will not be affected by three of the six surfaces in normally shaped rooms." And so it proved to be.



The first illustration compares the level increase at the input of the mixer from a trihedral PZM (PZM™) as compared to a regular single surface PZM™. The TDS measurement is from 200 Hz to 10,000. The upper trace is the PZM^3 and the lower trace of the regular PZM. Note approximately 10 dB of gain.

The second illustration is of the front-to-back ratio of the PZM^3, at least 20 dB. The curves were made by turning the back of the PZM^3 toward the speaker.

Even more impressive than these measurements is the listening quality. Gain jumps up and room influences lower dramatically. Placement of the microphone element is critical and without TDS-EFC measurements extremely difficult to do properly.

As a system for recording conferences in small to medium sized rooms, PZM^3 is without peer. Use of the PZM^3 away from large supporting surfaces merely acts as a highpass filter below the one-quarter frequency associated with the wavelength of the plates used to form the corner mounting.

Ken will be accepting a limited number of initial orders for the PZM^3 from experienced PZM users interested in helping us gather further field data on the optimum applications.

PZM™ IN THE COURTROOM

PAUL MORROW, consultant in Dallas, has designed a beautiful in-the-table model of the PZM™, the Model F, for several courtrooms that he is consulting for. Ken is manufacturing the Model F.

LETTER TO THE EDITOR

The biter got bit. We're always complaining about the lack of editorial control of material in current audio magazines. TERRY HOFFMAN, National Coordinator, Sound & Communications for Johnson Controls in Milwaukee, wrote us that we had neglected to do our duty with regard to GLEN BALLOU's opening statement in his TECH TOPIC, 'Constant Directivity Horns".

Glen said, "Constant Directivity Horns, i.e. horns whose polar patterns remain essentially the same at all frequencies, have recently come into being with the Altec Manta Ray series."

Terry is correct in chiding our editing. Terry writes:

> I must take exception to the opening paragraph of Glen Ballou's recent Tech Topic pertaining to constant directivity horns. The statement that these devices have recently come into being with the introduction of the Altec Manta-Ray series is certainly suspect with regard to its accuracy. Certainly, the Electro-Voice HR series "White" horns, which were introduced many years ago are capable of being placed in the same category as the new offering from Altec. From canvassing of large projects for our large branch offices, I am aware that a majority of audio consultants use the Manta-Ray and HR series horns interchangeably. This includes Bob Coffeen, Boner and Associates and many others, depending upon the driver requirements of the particular application.
>
> While I understand that the Tech Topics are not written by yourselves, I would urge you to edit the statements of opinion such as the one expressed by Mr. Ballou before they are allowed to be printed in the Tech Topic series. For all of my complaining, I would certainly admit that the subject article was indeed pertinent and useful for training purposes, and I would thank both Glen Ballou and Synergetic Audio Concepts for putting it in print.

MODIFICATION OF PEUTZ PROGRAM

Bill Raventos of Ivie has rearranged the Peutz articulation program into a format even more compatible to the IE-30 and IE-17 combination. The Ivie units allow relatively easy access to the four parameters required for the Peutz program - L_D, L_R, L_N and RT_{60}. Here's Bill's adaptation and comments.

> As I went through the program it seemed to me that there might be an easier way to manipulate the program on the calculator, and so I did a little re-writing. The program I've enclosed does, I believe, exactly the same mathematics; it's just that the format for manipulation is a little different. I note that it saves a few steps (Peutz's program is 130 steps and this one that I've done is only 100 steps. But the savings in steps really wasn't as important as making it a little easier (to my confusable mind) to enter and change data.
>
> To enter the initial data one simply enters the number and presses the appropriately labeled button. After all the data is entered press button A to run the program and display % Al_{cons}. At any time one wishes to change any part of the data, simply enter the new number and press the appropriately labeled button. Pressing the gold key and A will clear all registers and reset the program.
>
> On the whole, I don't suppose there's a whole lot of difference between the programs and any time saved; it just goes to show that one does tend to understand the working of the program better if he writes it or manipulates it to his own liking! The fact that it is relatively simple to measure the parameters necessary for calculating $\%Al_{cons}$ with this program using the IE-17A probably adds to my enthusiasm...

INSTRUCTIONS

1) Enter Program Card
2) Enter L_d, press B
3) Enter L_r, press C
4) Enter L_n, press D
5) Enter T_{60}, press E
6) press start, read $\%Al_{cons}$

To change any number, enter and then press appropriate button (B thru E). Pressing f A clears registers and resets program,

HP97

```
001  *LBLB
002  1
003  0
004  ÷
005  10^x
006  STOB
007  R/S
008  *LBLC
009  1
010  0
011  ÷
012  10^x
013  STOC
014  R/S
015  *LBLD
016  1
017  0
018  ÷
019  10^x
020  STOD
021  R/S
022  *LBLE
023  STOE
024  R/S
025  *LBLA
026  RCLE
027  1
028  2
029  ÷
030  LOG
031  2
032  ÷
033  CHS
034  STO5
035  RCLD
036  RCLC
037  1
038  0
039  x
040  RCLD
041  +
042  ÷
043  LOG
044  .
045  3
046  2
047  x
048  CHS
049  STO6
050  1
051  X≤Y?
052  STO6
053  RCL6
054  STx5
055  RCLC
056  RCLD
057  +
058  STO7
059  RCLB
060  1
061  6
062  x
063  RCL7
064  +
065  ÷
066  LOG
067  .
068  3
069  2
070  x
071  CHS
072  STO8
073  1
074  X≤Y?
075  STO8
076  RCL8
077  RCL5
078  +
079  RCL8
080  RCL5
081  x
082  -
083  2
084  x
085  CHS
086  10^x
087  .
088  6
089  1
090  5
091  +
092  1
093  6
094  6
095  x
096  RTN
097  *LBLa
098  CLRG
099  RTN
100  R/S
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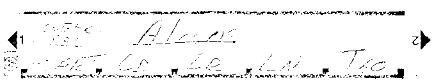
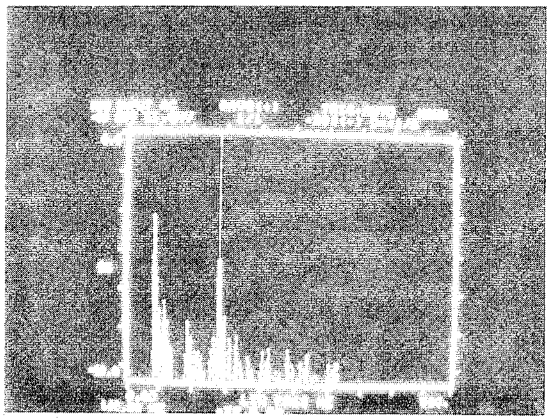


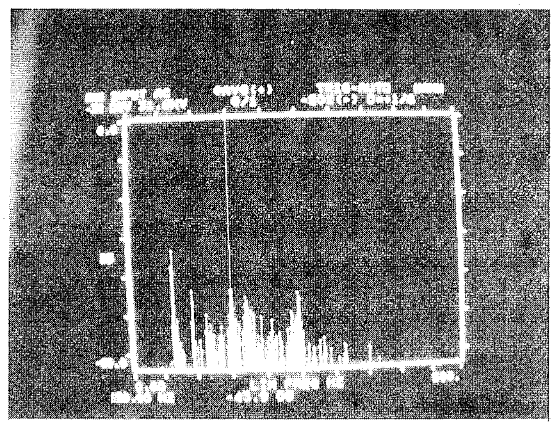
PRELIMINARY DATA ON STUDIOS AND CONTROL ROOMS

Syn-Aud-Con is, whenever the opportunity presents itself, measuring what are subjectively judged to be good studio recording setups. We attempt to measure with the test loudspeaker where the performer generated his sounds and the measuring microphone where the recording microphone had been. This allows us to approximate the initial time delay (ITD) generated by the performer in the studio itself. Our interest is twofold. First, we are interested in determining if the ITD of the studio can be reproduced in the control room by the control room monitor loudspeaker system. Secondly, we are looking for a correlation between setups to see if some optimum level, spacing, or other parameter appears in the data which would then allow us to plan the studio portion with greater accuracy.

We took the Anaheim class to see and measure a well-designed studio of what is called the "Hidley" type design.

The Anaheim class on a field trip.





This illustration shows the studio ETC curve glass doors closed. (Cursor on the glass doors) A "Hidley" type studio.

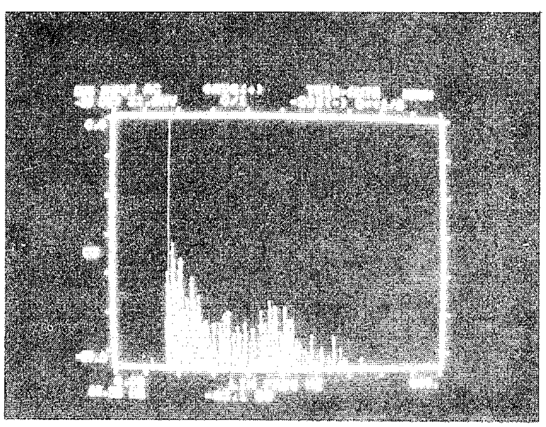
This illustration is of a very large studio (sound stage size) of the older conventional west coast pattern. What is of interest to us is that the ITDs of the first significant reflected spectrums are both 9.0 msec. (That is, 9.0 msec elapses between the arrival of the direct sound and the first significant reflection (cursor on the significant reflection in both cases). The level of the small room's first significant reflection relative to the direct sound level is -9 dB and for the large studio it is -10.5 dB.

When it is realized that one studio is approximately 8000 ft^3 in volume and the larger one approximately 300,000 ft^3, then the significance of these ITD's become worthy of further study.

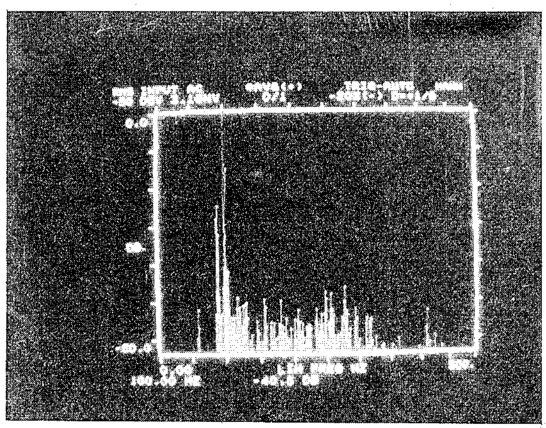
In the case of the larger studio, Goboes had been employed to provide earlier reflections to the musician and, most interestingly of all, the first significant (only about 10 dB less than the direct) reflection turned out to be the music stand holding the sheet music. (Cursor is on the reflection from the music stand.) Obviously this problem demands some attention.

What Happens in the Control Room?

That's a good question. In most cases of contemporary design, the studio's ITD has about the same chance as Custer at the Little Bighorn.



→→→The first control room illustrated is of the ETC of the monitor in the control room. Over 200 such "time smeared" control rooms have been built in recent years.



This illustration is from an LEDE™ control room →→→ with Time Aligned™ speaker system. The control room has an ITD 20 dB minimum depth and 15 msecs long. This control room could easily reproduce with a high degree of accuracy either of the two studio ITDs pictured above.

Preliminary Data... continued

Of very practical interest is the fact that these gross differences are clearly audible.

Conclusions

A coherent pattern appears to be developing with regard to what comtemporary musicians regard as a desirable acoustic environment. That it is temporally patterned rather than frequency patterned is worthy of a great deal of detailed study. The gross misdesign of a vast majority of hitherto well regarded control rooms raises both technical acoustical questions and reveals startling aural deficiencies that mixers have been willing to live with in the past.

HP 41C - A POWERFUL SYSTEM

Hewlett Packard's new HP 41C handheld calculator turns out to be an exceptionally powerful system. It is the first handheld calculator to offer full alphanumeric capability on *display* and keyboard. It uses HP's own segmented LCD readout, displays 12 characters at a time and allows 24 characters to be entered at one time with the display advancing through the string.

Our unit, purchased for $245 from Olympic Sales has 400 lines of program, or 64 data storage registers in any mix selected by the user. The HP 41C has four input/output (I/O) ports and up to four RAMs can be plugged in for 64 registers per Ram, or a total of 2000 lines, or 319 data storage registers. Since it already has 130 preprogrammed functions, we are talking about a powerful computer.

We ordered the plug-in magnetic card reader (list price $195) three plug-in RAMs ($45 each) and the thermal printer (list price $350). All but the printer have been delivered. A Wand for reading bar codes will be available in early 1980.

Using a 56 bit serial CPU that is similar to that used in their HP 97-67 units but with a faster clock and advanced software, the HP 41C is nearly twice as fast as its predecessors.

In 1973, Syn-Aud-Con purchased for $7600 an HP 9820 tabletop computer for use in the classes. The new HP 41C has every feature the 9820 had plus more preprogrammed functions, more memory, the plug-in Wand, and most unbelievable of all, continuous memory all powered by four throwaway N cells said to last for up to six months (not if you use the card reader steadily).

Any key on the HP 41C can be assigned in the "user mode" to be anything the user wishes. I currently have keys for:

TEF -- finds the full scale time for ETC plots on the FFT

Time -- gives the time in milliseconds for the apparent frequency reading of the FFT cursor

Travel - gives the distance in inches for the apparent frequency reading of the FFT cursor

Reflection -- computes the absorption coefficient for any given dB value of change in a reference level

Davis -- the complete Syn-Aud-Con sound system design program as re-written for alpha and sound by Bill Raventos

%AL -- the latest Peutz articulation loss of consonants percentage as calculated from L_D, L_R, L_N, and RT_{60} (Raventos version)

dB -- a decibel program for changes in dB-SPL with distance and/or power

I still have room for several more reasonably sized programs. All I do when I need one of these programs is turn the unit on, select user mode, and press the appropriate button. No cards to be read, pauses to be taken or any other interruption of my calculations.

Just to describe the features unique to the HP 41C would take the whole Newsletter. Many functions are called up by using alpha, such as putting the unit into the radian mode: XEQ (execute), alpha, RAD, alpha, and then RAD appears on the readout below the numbers or letters on a set of annunciators.

As if all this weren't full recompense for the long wait, it is rumored HP also is planning to introduce a desk top home computer with cathode ray tube and a full line of peripherals starting around $2500.

Make no mistake, the digital age is fully upon us now, not later. I suspect that the HP 41C or some variation thereof will soon be directly attached to the latest instrumentation.

NEW PROGRAMS FOR THE HP41C

It was Bill Raventos that showed up in Tustin in late July to show us the new HP 41C with card reader. In the meantime, Bill, now complete with printer, has generated a flood of beautifully thought out programs.

Closely following Bill's latest programs in the mail was RAY RAYBURN's use of the HP 41C as a word processor plus still another version of the sound system design program.

No wonder Syn-Aud-Con stays ahead. Our input sources are fabulous, to say the least.

The sound system design programs take several pages to reproduce and there are not enough owners of the HP 41C to justify printing it in the Newsletter; however, we will photocopy it for those who want to order it. You may order a copy of the Raventos program or Ray Rayburn's program for $2 or both for $3.00.

CROWN

Crown International is living up to their name. MAX W. SCHOLFIELD, President of Crown International, Inc., recently announced that nine Soviet technicians had successfully completed a special audio service and repair program conducted by Crown personnel.

This effort is part of the backup for the use of 50 Crown amplifiers in the Moscow World Trade Center during the 1980 Summer Olympics. These 50 amplifiers, capable of delivering over 10,000 watts of audio power, are expected to play a vital role in the 1980 Summer Olympics.

Having worked with sound systems in Russia in the late 1950s, we can appreciate a little of the impact Crown's technology is having on Soviet audio men. Russians don't buy outside their own sphere unless the outside product is exceptional. That made Crown a natural choice.

CONTROLLING ACOUSTIC SIZE IN SMALL ROOMS

First, let's define what size a space is in terms of acoustical parameters. We consider a space "acoustically" small whenever no boundary surface is further from a critical listener position than 25 msecs (no initial signal passing the listener, being reflected to the listener exceeds a total of 40 msec). Large spaces, then, exceed 20 msecs by a substantial margin. The usual "grey area", so common in all psychoacoustics, is in the time interval from 40 msec to approximately 80 msec (where time smear becomes echo to most listeners).

20 msecs represents an *upper limit* in the case of critical listening situations such as recording studio control rooms. This, in turn, suggests maximum dimensions of

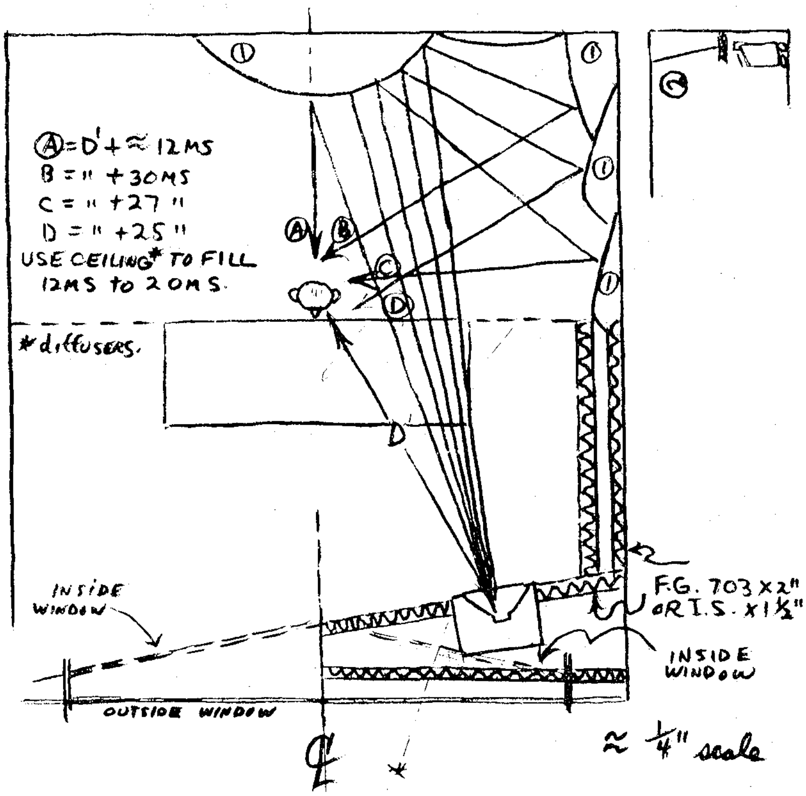
$$\frac{1130 \text{ ft}}{1 \text{ sec}} = \frac{x \text{ ft}}{.020 \text{ sec}}$$

$$x \text{ ft} = (1130)(.020) = 22.6 \text{ ft}$$

This is the *maximum* distance you would want a *reflective* surface from a critical listener (and one-half this distance is perhaps the *optimum* value.)

In a "Live end-Dead end" (LEDE) control room the dimensions of the "dead end" of the room (in front of the mixer) are not critical because they aren't as reflective, thus this criteria applies primarily to the back half of the control room where hard reflective surfaces are being employed.

BILL PUTNAM of United Recording Corp. has drawn up a rough sketch for obtaining a series of closely spaced (in time) reflections from the rear, side, and ceiling surfaces back to the mixer's position as well as indicating several methods of handling windows in the front wall.



Courtesy Bill Putnam, United Recording

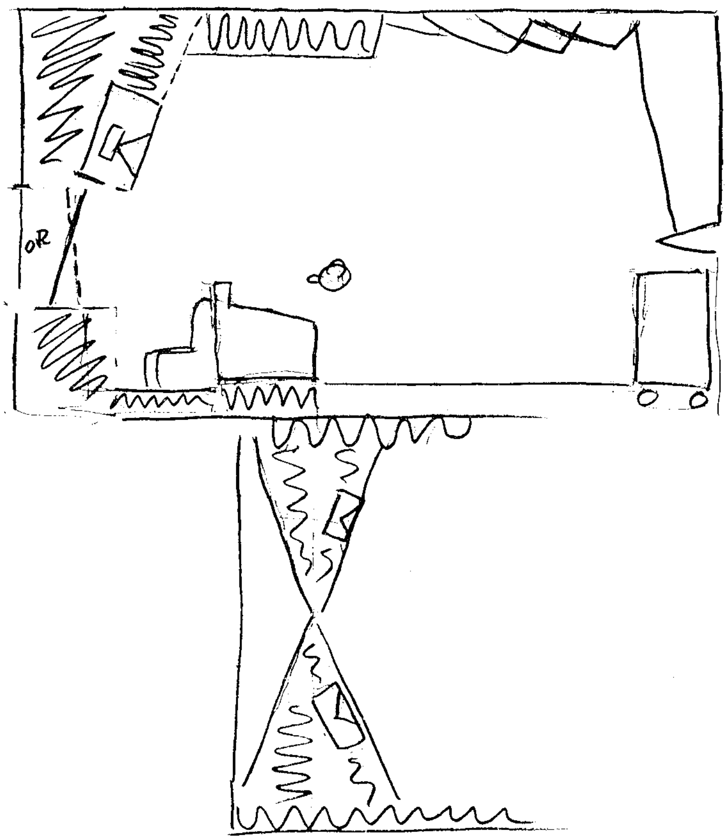
EXPLANATORY NOTES

(1) Polycylindrical diffuser or quasi polys would give max. diffusion but splays could be used less effectively. The polys *would not* be L.F. diaphragmatic absorbers as usually built in the conventional way using 1/4" plywood over random spaced bulkheads. They could be rigid and not affect the L.F. ā in the conventional way, if desired.

(2) In section the Polys could slant down providing a soffit for the machines and equipment as Putnam does in his Studio "A". Hard floor to rear of console and carpeted from front wall and under console.

Control room windows can be angled (preferably convex as viewed from the control room). The angle should insure that any reflections from the window strike the "dead" end side walls.

Continued next page



Courtesy Bill Putnam, United Recording

Obtaining a Semi-Reverberant Environment

As outlined in Tech Topic Vol 5 # 7, *PUTTING IT ALL TOGETHER IN THE CONTROL ROOM*, we can find the desired RT_{60} for a given volume and ceiling height by a convolution of the ΔdB equation of Peutz's

$$RT_{60} = \frac{.221\sqrt{V}}{\Delta dB \cdot h}$$

In using this equation, we have been choosing ΔdB values of 3.5 to 4.5 as good *initial* guesses. The opening between the live end and the dead end (the vertical plane at the mixer's position) is treated as a surface possessing 100% absorption. (That is, the dead end is treated in the same manner as a "coupled space" in large room acoustics.)

In this manner, our basic dimensions for the live end are dictated by our time interval constraints. 10'H, 15'D, 25'W are close to ideal.

This constitutes a ratio of 2:3:5, which while not a key parameter, it does not fall into a known harmful ratio. This yields an approximate volume in the area of 3,750 ft^3 and surface areas of approximately

Ceiling	= 375 ft^2
Floor	= 375 "
Side wall	= 150 "
Side wall	= 150 "
Rear wall	= 250 "
Opening to dead end	= 250 "

If our ΔdB is to be 3.5, for example, then

$$RT_{60} = \frac{.221\sqrt{3750}}{3.5(10)} = .39 \text{ secs}$$

And the total sabin value desired would be

$$S\bar{a} = \frac{.049V}{RT_{60}} = \frac{.049(3750)}{.39} = 471.15$$

Since the opening to the dead end has theoretically 250 sabins already, then the total sabins for the remainder of the surfaces should not exceed 221.15 ft^2.

Hardwood has an absorption coefficient of from .06 to .10, thus the ceiling, floor, side walls, and rear wall constitute, if left hard, 130 sa

$$221.15 - 130 = 91.15 \text{ discretionary } S\bar{a} \text{ to apply}$$

Ideally, any absorption of high value should be placed in the rear corners or can be placed on the floor under the mixer for quieting chair noise, etc.

What are the Benefits of Hardening One-Half the Control Room in the Rear of the Mixer's Position?

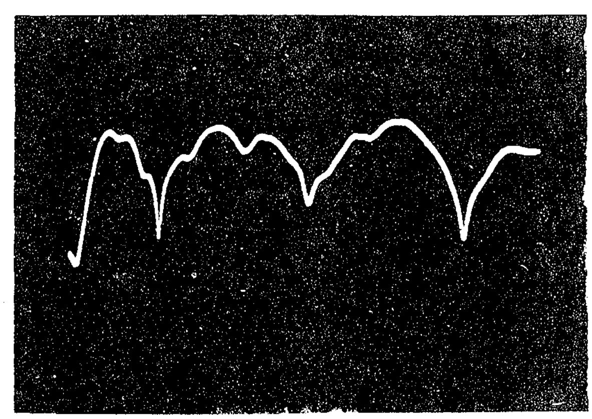
The most obvious one is to avoid broadband acoustic anomolies. The bandwidth of the anomolies is determined by the distance between the direct sound and the reflected sound at a given location (in this case, the mixer's ears). For example, if we were to examine the anomoly generated by the direct sound and a reflection returned directly from the rear wall, we can use

$$B.W. = \frac{V}{D}12$$

Where: BW is the bandwidth of the anomoly in Hz
D is the distance in inches
V is the velocity of sound in air (1130 ft/sec)

$$\text{Thus, } BW = \frac{1130}{(10' \times 12")} 12 = 113 \text{ Hz}$$

At a frequency of 500 Hz this is approximately a 1/3-octave bandwidth. At 1,000 Hz it is a 1/6 octave bandwidth, etc.



Total D = 6.78"

The major reason, however, is to establish the desired "initial time delay" gap (ITD). The rear wall, rear side walls and rear ceiling should, by means of diffusers, provide a series of spaced energy arrivals at the mixer's location that (1) provides an ITD slightly greater than the ITD in the studio (slightly is from 1 to 4 msec) (2) through diffusion lower the level of the total reflected field.

The second advantage, which also enhances the diffuseness of the total sound field is that using the hard rear wall *removes* any potential M_a factor *which in the case of physically small rooms is undesirable.*

In the example we are using here *if* the large rear wall were totally absorptive, $a_c = .99$, and our average absorption coefficient were $\bar{a} = .304$, then the M_a factor would have been

$$M_a = \frac{1 - (.304)}{1 - (.99)} = 69.6$$

This would mean that so far as the loudspeakers are concerned (if all the energy from the loudspeakers hit the rear wall first) this room has 70 times the average absorption. Even in the more likely case of only half the loudspeaker's energy hitting the rear wall first, the multiplier remains a very healthy 35.

To a talker in the middle of the room, it would seem on the "live" side but the loudspeaker would behave as if in an anechoic chamber.

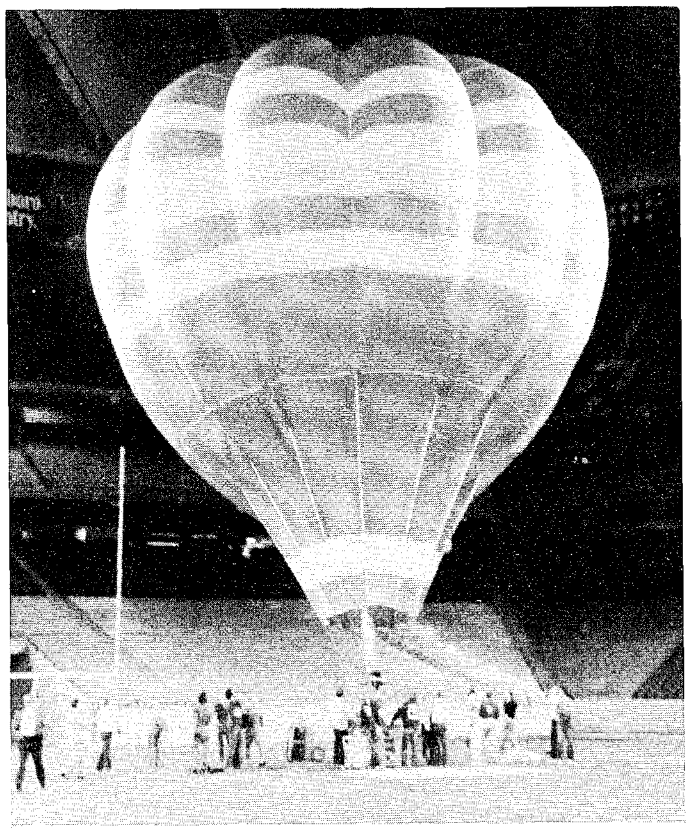
Some Conclusions

In physically small rooms (maximum reflection double path - 40 msec), several acoustic principles reverse as compared to physically large rooms.

1. In physically small rooms, every effort is made to *minimize* M_a. In physically large rooms, every effort is made to *maximize* M_a.
2. In physically small rooms, every effort is made to achieve *long* reflection paths. In physically large rooms, every effort is made to achieve short reflection paths. In both cases, desired reflection paths are equal in length.

The establishment of a diffuse semi-reverberant sound field is desirable in both cases. Acoustically, small rooms demand a different thought process on the part of the acoustician than do large rooms. Simple geometry or ray tracing is a useful technique in the small room whereas statistical analysis (the D_c concept) is invaluable in large rooms.

HOW TO SERVICE A LOUDSPEAKER ARRAY



Can you top this?

The hot air balloon is being readied to go up to try to service an inaccessible loudspeaker array (the attempt failed).

BILL McCALL of Shalco (representative in MI) sent us the picture some years ago in color.

The flight was a smooth one but the approach to the array proved impossible. As Bill says, "While the attempt was futile, no one can say it wasn't innovative."

This epic occurred inside the giant inflated dome in Pontiac, MI. Another reason not be shy when the owner fails to clearly indicate how you are to service a remote loudspeaker location.

I'm sure that the Silver Dome has solved this problem in the years intervening since this balloon flight, but it is obvious that it wasn't a simple solution.

ANNUAL PURCHASING DIRECTORY

We get calls asking us to help find manufacturers addresses and phone numbers. We are glad to help but we would also recommend that every sound contractor have on their shelf the Sound & Communications 1979 Blue Book - the unique Annual Purchasing Directory. $5.75 each. You will find all your audio industry manufacturers, their company personnel, reps, addresses and phone numbers. You shouldn't be without it. Write Sound Publishing Co., 156 East 37th St., New York, New York 10016.

Another publication that is broader in scope that you should have is the Electronic Industry Telephone Directory. It lists all manufacturers in electronics but just the name, address and phone number. You can write, Harris Publishing Company, 33140 Aurora Rd., Cleveland, OH 44139, 216-248-8540. I can't tell you the cost of the publication as our representative in San Francisco supplies our copy each year (Moulthrop Sales Inc).

MEASURING Q & C∠ WITH ETC

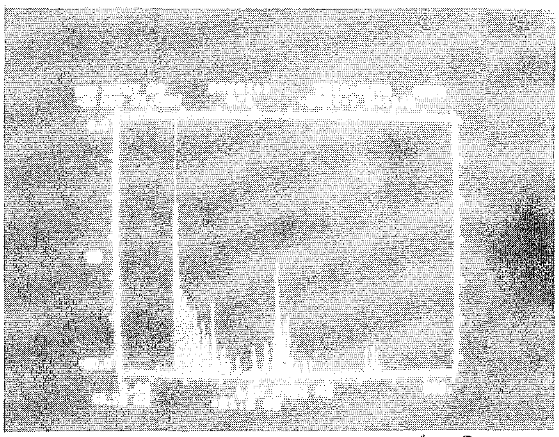
It is child's play to read total direct sound spectrum levels and maintain *equal distances* from a sound source while obtaining polar responses with Energy Time Curves (ETC), because the height of each line on an ETC display is the integrated energy density of the total spectrum it represents and each spectrum is separated in time on the horizontal scale.

In fact, only a single polar response need be obtained for a given loudspeaker and it can be band limited to fit the intended application (i.e., for heavy duty commercial sound horns) 500 to 5,000 Hz is a very adequate spectrum over which a single number Q value can be obtained from a single polar plot.

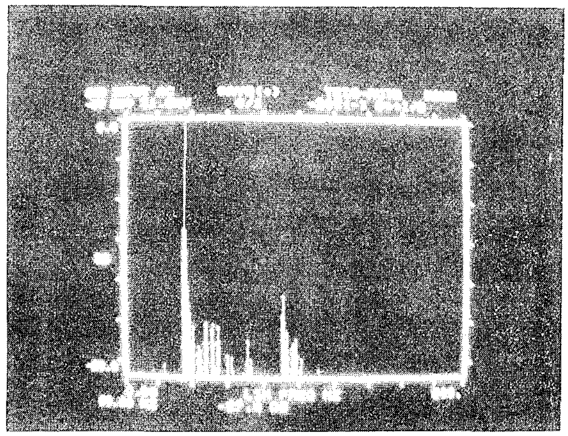
In the case where a very directional device is down -6 dB, say at 20° vertically and 40° horizontally, and at the next 10° increment is down 16 or more dB, and the remainder of the pattern also does so, it is relatively safe to use

$$Q = \frac{180}{\sin^{-1}\left(\sin\frac{\Theta}{2} \times \sin\frac{\Phi}{2}\right)}$$

Where Θ is the horizontal angle in degrees at the -6 dB point
Φ is the vertical angle in degrees at the -6 dB point



MANTA RAY ON AXIS 0dB REF



MANTA RAY INNER EDGE -5.6dB

We recently measured an Altec MantaRay directional control device and found that its direct sound spectrum is down over 40 dB at the sides and rear.

One remarkable aspect of these measurements is that you can't measure that kind of level change in an anechoic chamber (an anechoic chamber is limited to approximately 20 dB of discrimination against wall reflections - 99% absorption)

$$dB = 10\left(\frac{\ln(1-a)}{\ln 10}\right) \quad \text{or} \quad 10\left(\frac{\ln(1-.99)}{\ln 10}\right) = -20 \text{ dB}$$

The two illustrations show the principle of taking these measurements from the screen of the ETC. The FFT analyzer's cursor is set to the direct sound spectrum. Now, the screen reads relative levels directly and all that needs to be done as angles are changed is to be sure the direct sound spectrum is kept under the cursor by moving the microphone back and forth. So long as the direct sound remains under the cursor, the acoustic distance from the source to the microphone remains the same.

In addition to being a more accurate, higher resolution form of these measurements, it's also the easiest and *can be precisely duplicated in the field.*

CONTINENTAL VERSION OF D_C EQUATION

CLAUDE VENET, that one man French Foreign Legion (he's lived in England, the U.S. and now South America in addition to his native France) came up with the following interesting variation of the continental version of the critical distance equation

$$RT_{60} = \frac{(.013121)^2 QV}{(4D_{12})^2}$$

Where: RT_{60} is the reverberation time for 60 dB of decay in seconds

Q is the test loudspeaker's directivity factor (dimensionless)

V is the internal volume of the room in ft^3

D_{12} is the distance (on axis) from the test loudspeaker at which the sound is 12 dB *above* the steady reverberant sound field in the room

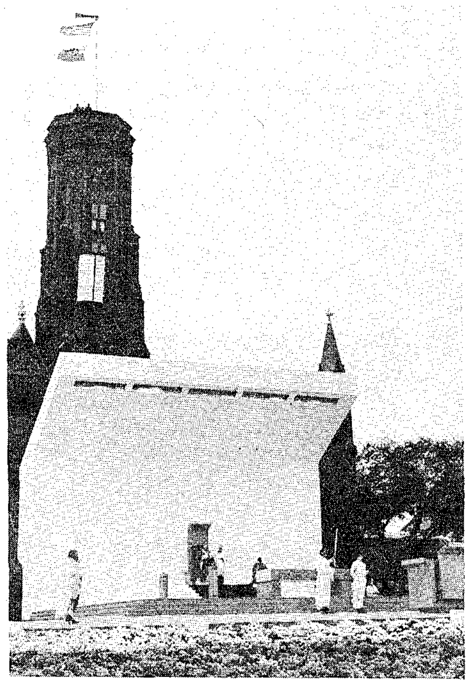
All that is needed to obtain as accurate an RT_{60} as any other device available is a loudspeaker with its Q available in octave bands, a sound level meter with an octave band filter, a noise generator and a tape measure.

Clever, these Frenchmen.

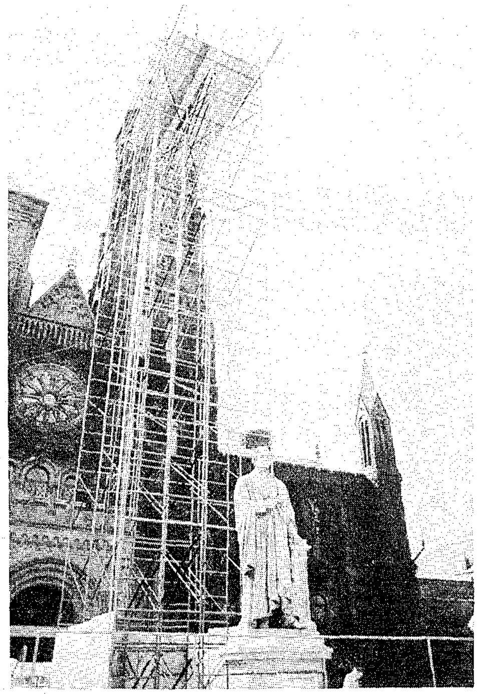
SOUND SYSTEM FOR THE POPE IN WASHINGTON

FARREL BECKER of Kensington, MD flew back to Washington, D.C. at the request of Syn-Aud-Con graduate, MIKE HOOVER at Audio Technical Services in order to assist in the set up and operation of the sound system for Pope John Paul's outdoor services there. (Farrel and Gina Becker are spending this Fall and Winter working at Syn-Aud-Con's California office.)

In the first photograph (taken from the mixing console position)→→→ the array of EV horns stacked side-by-side can be seen above and behind the altar area.

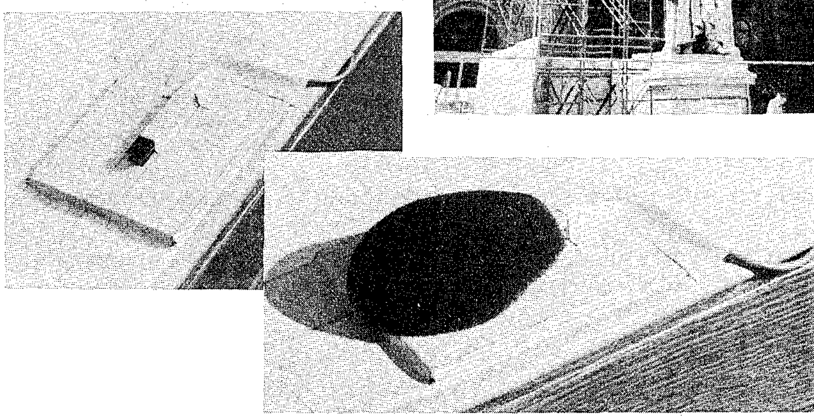


This photograph is taken at the base of the tower on which the main array is assembled.



Much to Farrel's delight, the statue at the base of the tower is Smithsonian's first secretary, Joseph Henry of "Henry effect ne: Haas Effect".

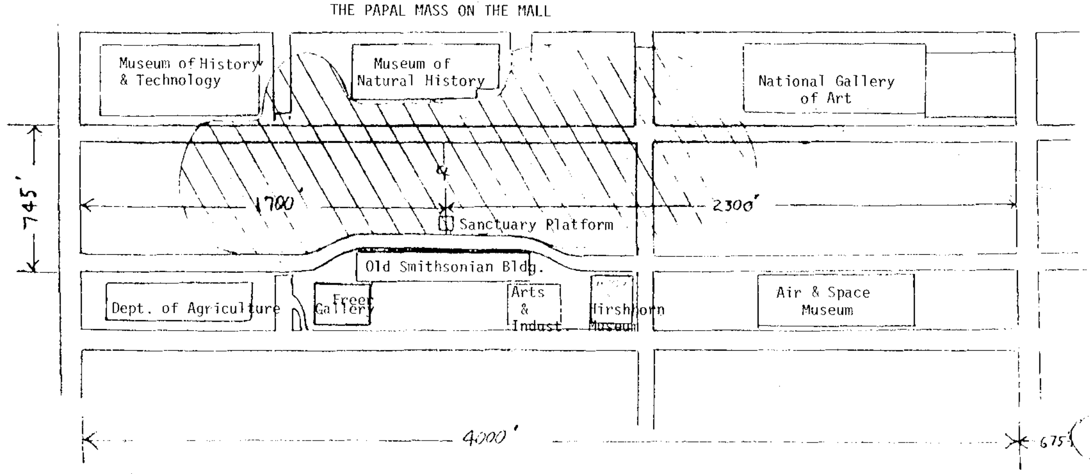
It was Joseph Henry in 1847 who first observed and wrote a paper on the inability of the human listener to detect reflected signals delayed approximately 20 to 30 msec. Farrel felt that having Joseph Henry oversee the elaborate electronic time delays being installed for the arrays used for the coverage of the audience attending this service was especially fitting and proper.



This photograph shows the altar PZM™ with and without windscreen. PZMs™ were used for the choir as well. The wind noise heard over the telecast was from the conventional microphones used at the pulpit.

Farrel wrote up the technical aspects of the sound system:

THE PAPAL MASS ON THE MALL



Although acoustical consultants were hired by the architect, their sole purpose was to select a contractor to do the job. Selection was based on how the contractor would handle the job, equipment available and overall competence. Price bid was not taken into account. Audio Technical Services of Vienna, VA was awarded the contract. Owner MIKE HOOVER spent the next 3 weeks designing and assembling the system.

The system was required to be totally redundant. Any piece of gear that was not duplicated was rigged to be by-passed. The area to be covered was approximately 745 feet by 4,000 ft. This coverage was accomplished with 5 speaker towers, each 65 feet in height. The main tower, located in the middle of the 4,000 foot side, consisted of 4 EV TL606Q bass cabinets and 12 EV HR 4090 horns. See photo # 1.

continued next page

Sound System for the Pope, continued

The next tower in either direction (towers 2 & 3) consisted of 2 TL606Q and 8 HR 4090.

The outermost towers (towers 4 & 5) consisted of 2 Klipsch MWM bass cabinets and 2 Klipsch MSM midrange horns. The equalizers, crossovers, and power amplifiers for the remote towers (2-5) were located in racks at the base of each tower. The electronics for the main tower, as well as the time delay units for the remote towers were housed in a mobile office trailer located behind the sanctuary platform. This trailer served as a base of operations during the week long set up. During the ceremony the trailer and all four remote towers were manned and connected by intercom. Any equipment failures (and there were none) could be taken care of instantly. The main mixing consoles were a Yamaha PM-1000-16 and an API 16x4.

There were two microphones side-by-side at each of the 5 locations on the sanctuary platform. One microphone from each location was fed to each console for redundancy. Four Pressure Zone Microphones were used on the 450 voice choir, one microphone on the band, and one microphone and a direct output for the organ. Each console fed the speaker towers, the front precedence speakers, monitors, and the television pool separately. The API console was used for the vocal mix and the Yamaha for the music mix. Both consoles were set up to do both mixes instantly in case the other one failed.

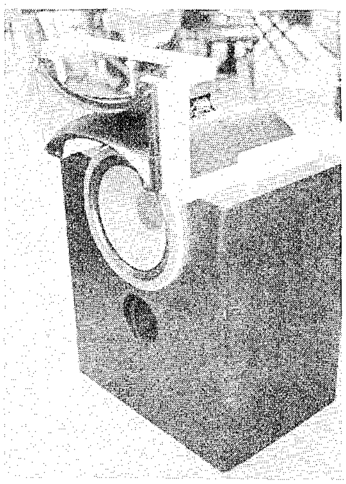
Eight Klipsch Heresy speakers were placed along the front edge of the Sanctuary platform as precedence speakers at the edge of the flowers in Photo # 1. With Mr. Henry looking on, these speakers were time delayed to be 20 msec ahead of the main array. The sound for those seated near the platform then seemed to come from the platform instead of 65 feet overhead.

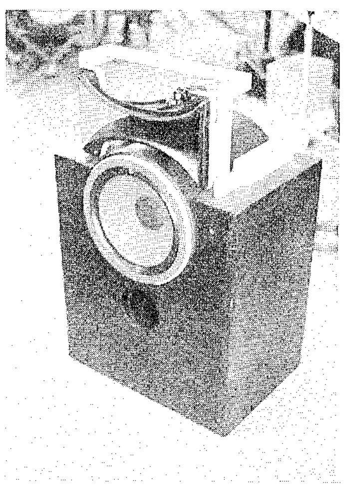
The ceremony went very smoothly and reports from various listeners afterward indicated that coverage was excellent and very articulate from the Washington Monument to the U.S. Capitol building, a distance of one and a half miles.

Of the eleven people involved in the set up and operation of the system, six were Syn-Aud-Con graduates. They are MIKE HOOVER, SKIP MARUT, BRIAN HOOVER, RALPH HOOVER, ALAN PERRY and myself, FARREL BECKER.

"SLED TESTS", TIME ALIGN™, AND BERNIE'S EARS

One of the frequently asked questions in recent classes is, "how audible is a mis-time aligned loudspeaker?". Manny Mohageri, president of Emilar, built us a special loudspeaker system with a calibrated moveable high frequency horn assembly.





Special loudspeaker system built by Emilar with a calibrated moveable high frequency horn assembly.

Ken Wahrenbrock built up a special switchbox for this loudspeaker which allows us to quickly select either the L.F. driver, the H.F. driver, or both, while keeping everything properly terminated (especially the crossover network output legs).

By placing the FFT cursor on the woofer alone, for example, and then adjusting the position of the H.F. horn until its energy falls directly under the cursor, exact alignment can be achieved. As two drivers come into exact alignment the level can be seen to raise 6 dB on the screen.

Enter Bernie Cahill of Rauland Borg Corp. (TEF™ Workshop). Bernie is one of those jewels in the firmament who's not afraid of the new and difficult, while rich with a lifetime of very practical audio experience.

Bernie stood in front of the loudspeaker and while it was being fed a pink noise signal, told us when to stop moving the high frequency unit. Each time Bernie's ears and the ETC agreed (that's Bernie in the middle of the photograph). Many of the others in the class who tried it found that there is a short "learning" period and then, like so many effects of this sort, the unaligned positions become intolerable (catastrophe effect). Ed Long looked on with pleasure inasmuch as he has patiently and persistently said this for lo these many years.

Our thirty-plus years in audio has taught us an important lesson: If you *learn* how to listen, loudspeakers sound worse and worse with time. And the live instrument sounds better and better as you train your ears in its presence. Each improvement helps but in all honesty, we should keep constantly in mind that we're still working on correcting only the grosser distortions..

It's measuring sessions like the recent TEF™ Workshop that provide the experiences that can't be learned from a book - but then that's why you come to a Syn-Aud-Con class in the first place, and second place, and third place.

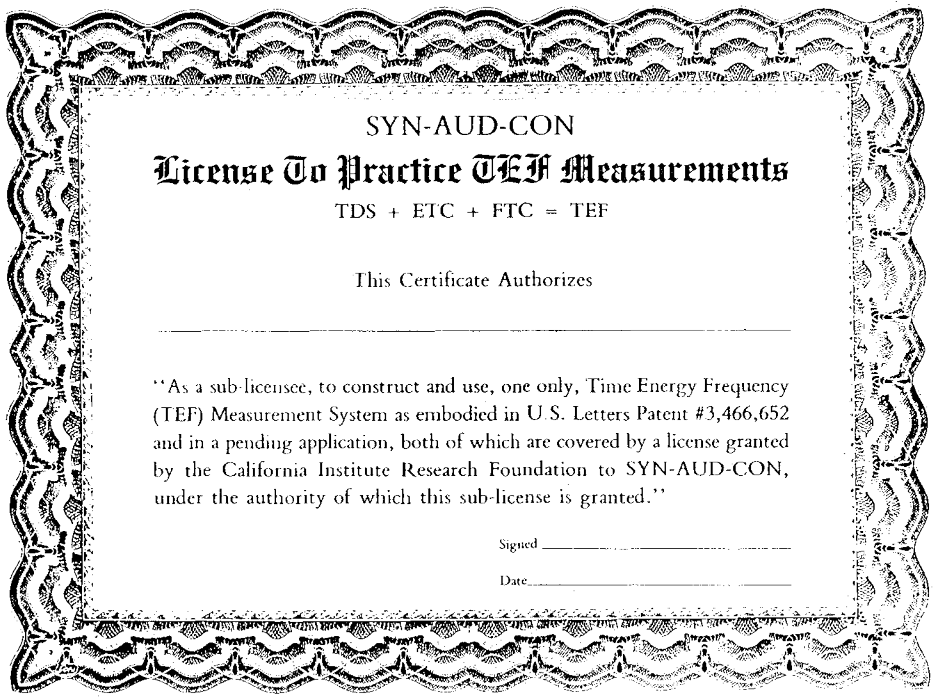
TEF™ LICENSEES

Like the original TDS licensing which took practical experience with the measurement apparatus and techniques, so the TEF™ licenses are applied by those who have had the opportunity to see the equipment in action.

Since our July Newsletter, TEF™ licensees have nearly doubled.

Just a few of the measurements we are demonstrating in the classes using our TEF™ apparatus are:

1. Alignment of multiple loudspeaker arrays
2. Measurement of Q and C_L to greater accuracy than any previous method.
3. Measurement of the absorption coefficient of every surface in a room with a single sweep of the ETC analyzer (using the 1/r compensator as described in Tech Topic Vol 7 #1)
4. Direct measurement of the mean free path
5. Direct viewing of the decay rate for each major axis in the room
6. Inherent "time smear" in large horns if not carefully designed to avoid it
7. Listening to the reflections in a room as tonal components where pitch is distance (higher pitch is greater distance of travel - described in Tech Topic Vol 7 #1)


SYN-AUD-CON

License To Practice TEF Measurements

TDS + ETC + FTC = TEF

This Certificate Authorizes

____

"As a sub-licensee, to construct and use, one only, Time Energy Frequency (TEF) Measurement System as embodied in U.S. Letters Patent #3,466,652 and in a pending application, both of which are covered by a license granted by the California Institute Research Foundation to SYN-AUD-CON, under the authority of which this sub-license is granted."

Signed ______________________

Date ______________________

All of this, in addition to the real time ability to detect where each reflected spectrum came from (direction of arrival), the distance it came (travel distance), and the time it took (travel time) plus the ability to tune in on any reflection observed to be significant without having to "search" for it.

Not in our wildest dreams of just seven years ago did we imagine tools as accurate, versatile and useful as TEF™. "In the land of the blind, the one-eyed man is king" describes the position you are in when you have TEF™ available and your competition does not.

While the original TDS license essentially covered only what we now call EFC measurements, the TEF™ license includes all present forms of measurement under Dick Heyser and Cal Tech's patents and patents pending on this process. So while the price seems higher, it actually covers access to so much more technology than before as to be one of the few genuine bargains in science today.

Access to these special licenses to practice TEF™ individually are due to be issued for only about one-and-a-half to two years maximum. After that, it is expected that there will be a dedicated instrument on the market and those desiring to investigate TEF™ will have to buy that instrument *unless* he possesses an individual TEF™ license allowing him to assemble his own system.

Original TDS licensees can advance to a TEF™ license for $400. A new licensee pays $500 for the license. All checks are made to Syn-Aud-Con.

TEF™ LICENSEES

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FILMSWAYS-HEIDER RECORDING
1604 CAHUENGA BLVD
LOS ANGELES, CA 90028

HOW TO STUDY

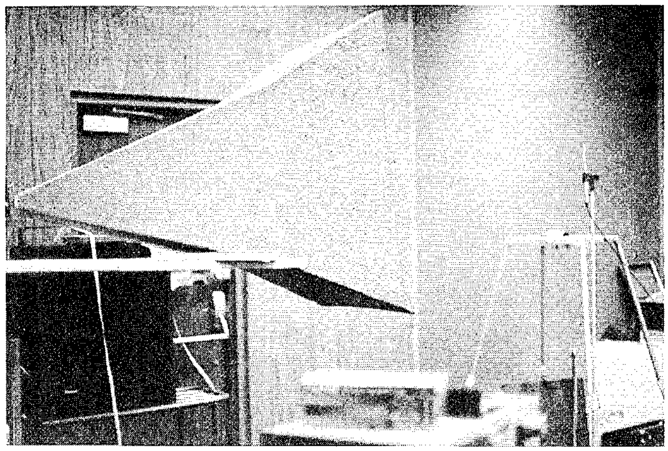
Errors like straws upon
the surface flow

He who would search for
pearls must dive below

John Dryden
1631-1701

A VISIT TO THE "HAUNTED" ETC HOUSE

You don't believe in ghosts? The September class sure thought we had one while measuring a highly directional horn (the MantaRay) with ETC. The set up is shown here→→→→



A spectral line appeared on the ETC screen *before* the direct sound. If the FFT cursor was set on this line and the TDS analyzer tuned to it, the EFC display showed no signal. Move the microphone closer to the wall and the ghost image moved closer to zero time delay (at the wall it disappeared) Move the microphone further from the wall and the ghost image approached the direct sound line.

Put a small loudspeaker such as a Visonik in place of the highly directional device (Q over 50) and no ghost image appeared at any setting of the controls. Place the Visonik *in the large horn near the throat of the MantaRay and the ghost image reappeares on the screen.*

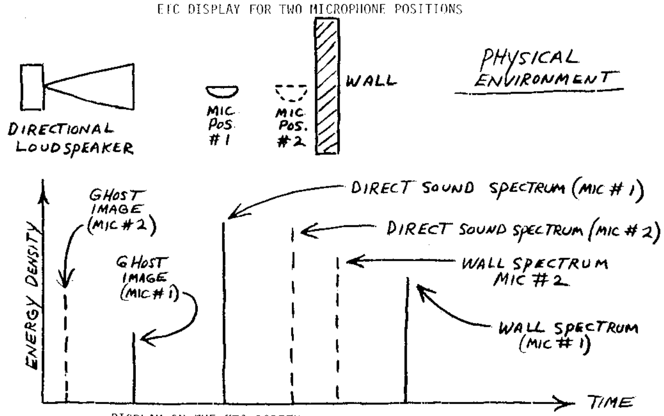
At the special TEF™ class in October we duplicated this effect in a different space with the same horn. Dick Heyser was present and was as successful as the rest of us in deciphering it.

It is suspected that we were looking at some negative frequency image from the TDS that passed to the ETC. The clue that it was not a real signal was its reverse time direction with microphone movement and its failure to appear on the TDS screen as an EFC signal.

The pictures below illustrate what a Syn-Aud-Con class looks like during a hands on session with a ghost image. "Hands on" seems to apply to discussions as well as measurements. We realized toward the end of the evening's measurement session that Masaaki Nakajima, Toa-Japan, had not said a word but had observed it all. We asked him what he thought about it. He said, "I think there are too many people making too much noise." We realized that Masaaki isn't used to seeing everyone talking at once to "honorable teacher"-- with raised voices.

This "adventure" kept the September class busy late one evening. It is easy to see the involvement of the people working on the project.

ETC DISPLAY FOR TWO MICROPHONE POSITIONS



DISPLAY ON THE ETC SCREEN

TESTS OF FINAL SBA PROTOTYPE

On the one-hundredth anniversary of Edison's perfection of the light bulb and his development of the electric power system for the lights, it still ranks as one of the truly great systems engineering feats of modern times.

Edison achieved such success because he turned from designing individual components for the telegraph industry and undertook the most arduous venture of his career - the design and integration of a large number of discrete components into a coherent highly technical system, the electric power and light system.

Sound system manufacturers tend to the design of eclectic components and leave system design to the individual entrepreneural contractors in the field. Thanks the efforts of less than three manufacturers and Syn-Aud-Con, there are a surprising number of systems-educated contractors available today, but not nearly enough to handle the ocean of daily sound system users.

It is the small system user in the hundreds of thousands that suffer through the "packaged components system", the parts house catalog wish-book , and other disastrous attempts to serve his need.

J W Davis Company of Dallas, Texas has served this mass user market through parts distributors and selected sound contractors. They have developed a very low cost collection of reliable components for their customers. Serving this marketplace made them all too aware of the need for an engineered systems approach, divorced from the complexities of the large expensive sytems.

J W Davis Company's answer to this challenge has been to bring to production Dick Heyser's remarkable invention called Signal Biasing Amplification (SBA).

SBA NOW READY FOR SALE

We have just finished testing the final prototype SBA Master, Remote, and supplementary power supply units, along with four $3.50 loudspeakers in plastic baffles.

We were playing back music at a level of 88 dB-SPL at 1 meter with clarity and without hum, noise, or distortion. No conventional amplifier could have made this inexpensive cone loudspeaker sound this clean.

Those of you that heard earlier demonstrations of SBA are in for a real but pleasant surprise. We are now hearing much higher levels at much lower distortion. Freedom of this system from extraneous outside interference is exceptional.

The J W Davis Company is to be congratulated on the really fine job they have done in tooling up to produce this system. We sincerely believe that the idea of the 70 volt distribution for background music and paging in low level sound systems now has all the finesse of installing gas lights in a new building instead of electric lights.

SBA FEATURES REVIEWED

The SBA system will handle from one loudspeaker to 100,000 loudspeakers with three simple interrelated system units. A Master unit (more than one might be needed for loudspeaker requirements in the thousands of units), remote units at each loudspeaker, and supplementary dc power supplies about every 15 loudspkeakers. SBA *is not* the inefficient idea of a power amplifier at each loudspeaker but a sophisticated distribution of the power needed in an unusually efficient engineering technique that literally stores power until needed in the loudspeakers themselves in the form of a dc offset.

Anyone who has need of a low level distributed sound *system* and wants demonstrably superior performance at a remarkable savings in total cost (when engineering and maintenance are accounted for) should hasten to contact Chappie Chapman, J W Davis Co., P O Box 26177, 3215 Canton St., Dallas, TX 75226. 214-651-7341. Note that this is a new address for the J. W. Davis Company.

CROWN RTA-2 AVAILABLE DIRECT FROM CROWN

As of August 1, 1979, Crown's RTA-2 real time analyzer became available as a direct sale component from Crown.

Just a reminder: The RTA-2 features

60 dB of dynamic range (on screen)
16 Hz to 20K Hz bandwidth
Both 1/3-octave and 1/1-octave display

It now costs $2195.

Crown is making available with these units the AKG C451E microphone with the CK22 capsule and the SA15-1 stand (normally $300) for only $100 when purchased with the RTA-2.

The portable case is now $100 and the shipping case $120.

As if that's not enough, all prepaid orders may deduct 6%.

If you, for example, pick up the entire deal

RTA-2	$2195.
Microphone and stand	100.
Portable case	100.
Shipping case	120.
	$2515.00
less 6%	150.90
	$2364.10

This may be the time to pick up that second real time you have always needed in the service shop, lab, or field truck.

SOUND SYSTEM COSTING FORM

One aspect of commercial sound work that is always of real interest to Syn-Aud-Con is how to correctly figure system costs. Without accurate costing information it is not possible to calculate a reliable but justifiable profit. MARK ROGERS of Intermountain Sound in Nampa, Idaho gave us his adaptation of a sheet we have in the black Lab Manual. We felt he had done an excellent job of organizing and recording needed data. Perhaps his form can aid you in the design of your accounting method.

ITEM	LABOR SHP	CHK	FLD	QUAN	✓	MODEL NO.	DESCRIPTION	UNIT COST	✓	EXTENSION
1										
2										
3										
4										
5										
6										
7										
8										
9										
10										
11										
12										
13										
14										
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23										
24										
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26										
27										
28										
29										
30										
31										
32										
33										

INTERMOUNTAIN SOUND

WORK SHEET ____ OF ____ DATE ____

SYSTEM ____

MAN TO SEE ____

TELEPHONE ____

ESTIMATOR ____

LADDER HEIGHT ____

SCAFFOLD HEIGHT ____

- TOTAL ESTIMATE
- JOB FACTOR - %
- TOTAL HOURS

	RATE	LABOR COST
FIELD		
CHECKOUT		
SHOP		
ENGINEERING		
OTHER		
TOTAL LABOR		

- MATERIAL COST
- USE TAX
- HARDWARE
- FREIGHT
- % MARKUP
- TOTAL MATERIAL
- TOTAL LABOR
- SUBLET COSTS
- BOND & PERMITS
- TRAVEL
- PER DIEM
- SALES TAX
- END SUM

ADD #	DEDUCT #	DESCRIPTION	ITEMS #:	COST + %	LABOR	TOTAL	NEW SUM

PROGRESSIVE BILLINGS

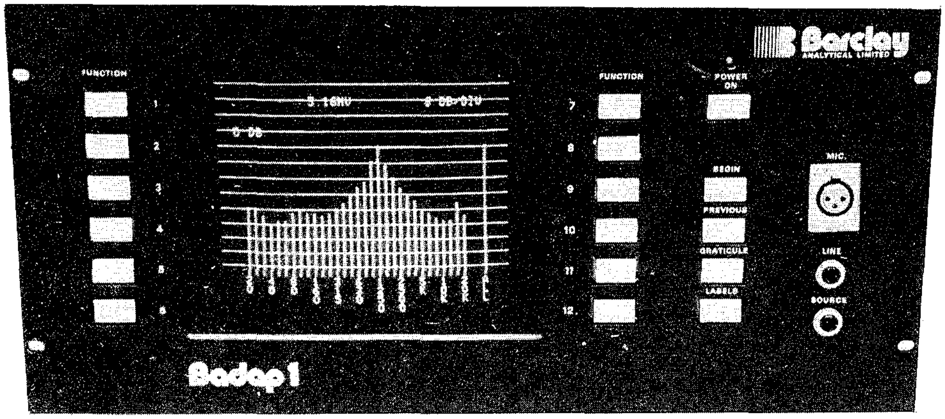
DATE	INVOICE #	% MAT	% LAB	TOTAL	BALANCE

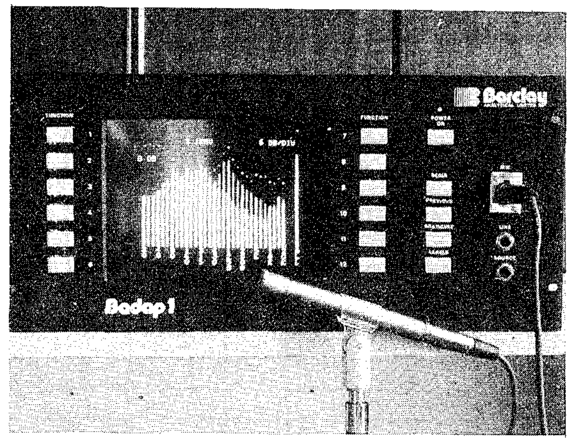
BADAP IMPROVEMENTS

The Badap 1™ Audio Micro Computer is now in full production.

We are told that a majority of early users have been ordering an optional RT_{60} program as well. As this new product reaches "in stock" status, Barclay is planning to show in November at the New York AES a new stereo analyzer.

The stereo analyzer provides two full spectrum readings at the same time with both peak and average readings for each.





The display can be switched to show either L and R, or L+R and L-R for disc mastering. The stereo analyzer allows two different microphone positions to be observed at the same time during equalization, i.e., one microphone well into the direct sound field and a second microphone well into the reverberant sound field.

Obviously, critical distance measurements would be a snap with this analyzer. The photographs show the latest production examples of the Badap I.

These units have to be seen "in the flesh" to be appreciated - the visual impact of the multicolor, multi-formatted display. We're not surprised that its initial reception has been enthusiastic. You can expect to hear a great deal from Barclay Analytical in the next few years. They are not ready to rest on their laurels yet.

PHASE & TIME

To find the "in-phase" and "out of phase" frequencies associated with time delay or displacement distance:

$$f_{360} = \frac{1}{T_D}$$

Where: f_{360} is the in phase frequency in Hz

T_D is the time delay in seconds

$$T_D = \left(\frac{1 \text{ secs}}{1130'/\text{sec}} \cdot \frac{1 \text{ ft}}{12''}\right)(\text{Displ. in inches})$$

In-phase frequencies = $f_{360} + f_{360} + \ldots$

Out-of-phase Frequencies

$$f_{180} = 0.5 f_{360}$$

Where: f_{180} is the out-of-phase frequency in Hz

Out-of-phase frequencies = $f_{180} + f_{360} + f_{360} + \ldots$

Phase Delay θ

$$\theta = \frac{T_D}{T_P}(360)$$

Where: T_P is the time period of $f \left(T_P = \frac{1}{f}\right)$

Thus $T_D = \frac{T_P \theta}{360}$; $T_P = \frac{T_D(360)}{\theta}$

continued on the next page....

PHASE & TIME, continued

Examples:

The acoustic center of two loudspeakers is 10" apart on axis (one is behind the other). What is the time delay between them? At what frequency is the first notch? What are the first three "in-phase" frequencies? What is the phase delay at 2,000 Hz?

$$T_D = \left(\frac{1}{1130} \cdot \frac{1}{12}\right)(10") = 737 \text{ usecs}$$

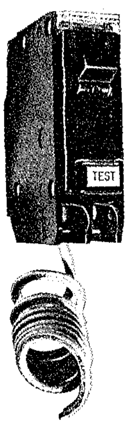
$$f_{360} = \frac{1}{(.000737)} = 1356 \text{ Hz} \qquad f_{180} = 0.5\left(\frac{1}{.000737}\right) = 678 \text{ Hz}$$

First three in-phase frequencies are 1356, 2712, 4068 Hz

$$\Theta = \left(\frac{.000737}{.0005}\right)(360) = 530.64^{\circ}$$

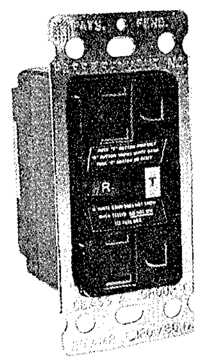
GROUND FAULT CIRCUIT INTERRUPTER (GFI)

In our classes we discuss the need for ground fault circuit interrupter especially the Hubbel, model GFP 115 that we carry with us. Building such devices into the master power systems often supplied to entertainment groups can literally be a life-saver. We recommend their use as one more protection you can provide the end user.



The *indoor circuit breaker type* ($40 to $45) installs in your breaker box and takes the same space as a conventional 120-volt breaker. It protects one entire circuit from current leakage.

The *indoor plug-in type* ($50 to $60) provides ground fault protection on any circuit. Plug it into nearest receptacle where you'll be working, then plug appliance into it, and you're safe.



The *wall outlet type* ($40 to $45) replaces a wall receptacle. It provides protection from ground faults at that receptacle only, or all receptacles wired into the same circuit.

How does a GFI work?

The GFI continuously monitors the current in the two conductors of the circuit—the phase or "hot" line and the "neutral" line. These two currents should always be equal. If the GFI senses a difference of five milliamperes (5/1000 amp), it assumes the difference is a ground fault (current is leaking) and automatically trips the circuit. Power is interrupted within 1/40 of a second or less. This is fast enough to prevent injury to anyone in normal health.

Any ground fault is dangerous. Even if you are using a faulty electrical tool with a grounding wire in good condition, you aren't completely safe. That third wire may be carrying off most of the leaking current, but should you touch a good ground, part of that leaking current could go through your body as well.

How much current is dangerous?

The amount of current that can pass through your body without doing severe harm is limited by individual resistance. But the average effects are as follows:

5 MA (milliamperes) to 15 MA: Local muscle contractions prevent the victim from freeing himself from the appliance or wire.

30 MA to 200 MA: This amount of current results in difficult breathing, unconsciousness, and ventricular fibrillation of the heart.

Over 200 MA: The victim suffers severe burns and muscular contractions; his heart fibrillates and may stop, resulting in death. (It takes about 200 MA on a normal household circuit to light a 25-watt bulb.)

THOMAS KING, Walt Disney World engineer, uses a GFI test instrument by ITE Imperial Corp - Ground Fault Tester, Model GFT 200. The address on the label is: ITE Datametrics, 340 Fordham Rd., Wilmington, MA 01887. Tom writes,

> There are several such instruments on the market and like audio equipment, there is a significant percentage of "duds". I suggest that when considering other types that you discount those pocket carried, plug-in, quick-testers with impressive lights and buttons. The only way to effectively test a GFI is to determine at what current the unit will trip and also what leakage to ground is already on the circuit prior to the connection of equipment suspected to be defective. The above described equipment (the Model GFT 200) does this quite well.
>
> The ITE instrument does draw about 1.5 ma of current (indicated on meter) even though there is no ground fault on the circuit. This is insignificant and will just have to be lived with.
>
> A good item to be carried around is an extension cord with a GFI receptacle mounted in a 4" x 4" "quad" box. By using the square-D "quick guard" receptacle and an additional duplex receptacle, you can have a 3 GFI protected sockets neatly packaged.
>
> General Electric sells a portable GFI (Cat. #TGTRP3) which also detects polarity reversal - this is a good device for demonstration. An industrial construction site type is manufactured by Hubbell. I doubt that this would be in line with your needs, but might be mentioned for those desiring protection at rock festivals in open fields.
>
> I am glad to know that Syn-Aud-Con is incorporating this subject into its training curriculum. It is only with the help of expert professionals like you and your staff that the true value of GFI use in the audio field can be advanced.

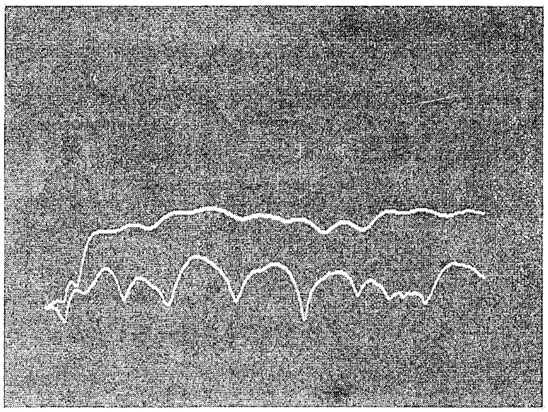
TDS TESTS OF SONEX ACOUSTIC MATERIAL

Illbruck/USA of Minneapolis sent us samples of Sonex acoustic panels. ED BANNON, chief engineer at One Step Up Recording Studio in Los Angeles, introduced us to the panels.

The test panels are approximately 4'x4' and test thickness included 4", 3" and 2" samples. Two patterns were included in our samples. One pattern we call the *indented pattern* as it appears to the eye as if a giant press caused the indentation seen.

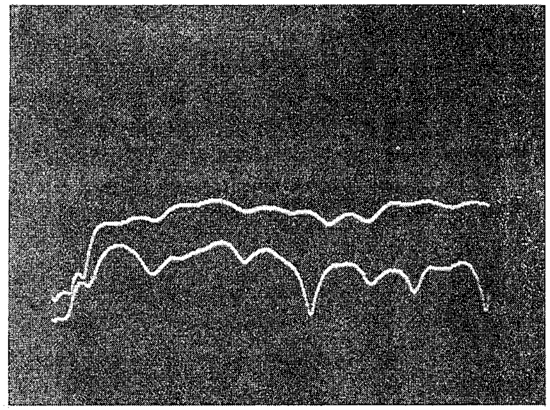
The second pattern resembles models, anechoic chamber wedges and we call it the *wedge pattern*.

In each photograph, the top trace is the calibration reflection off of our 4'x4' plywood backboard used as a mounting barrier for the test material samples. The sweep is, horizontally, 0 to 10,000 Hz linear scale. Vertically every 7mm equals 10 dB (7mm is approximately ¼").



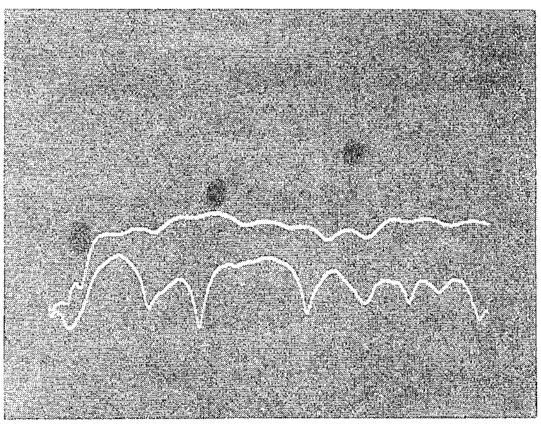
4" Indented Pattern

The minimum separation or absorption is 10 dB (a = 0.90) with broad areas of 20 dB (a = 0.99)



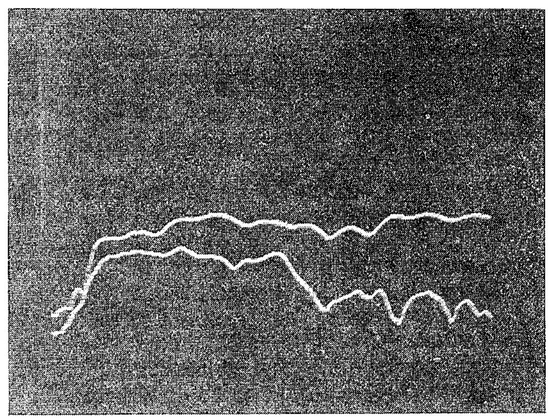
4" Wedge Pattern

A slightly smoother reflection after absorption indicating the wedges apparently have a functional reason for existence.



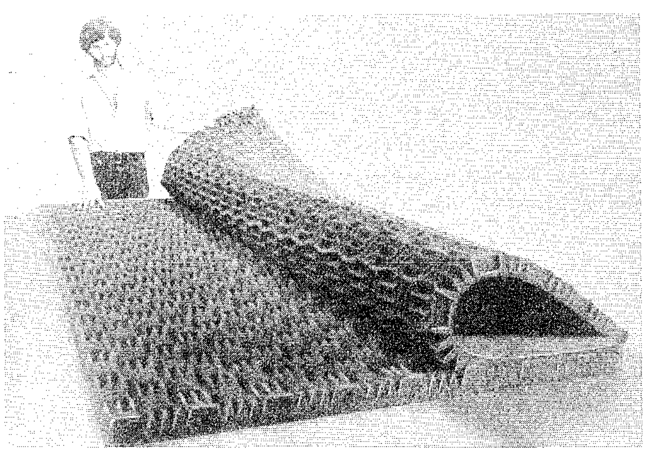
3" Indented Pattern

Slightly (very slight) less absorption

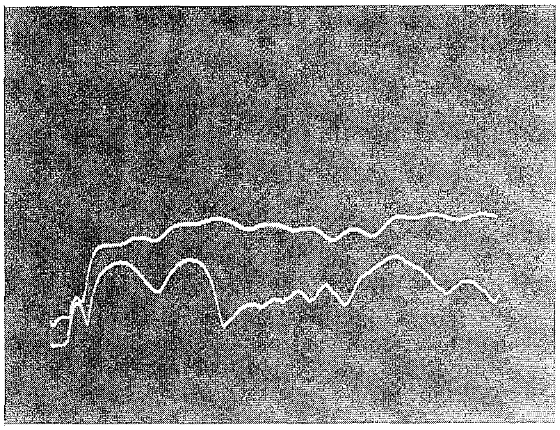


3" Wedge Pattern

We feel this to be a striking feature, frequency selective with greatly increased absorption from 5000 Hz up We will be investigating the mechanism causing this effect in the future, but in the meantime, it offers fascinating possibilities for interspersing panels of differing depths for increased diffusion, as well as absorption

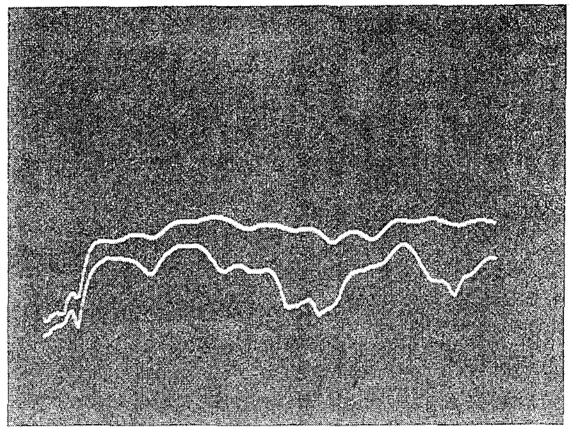


SONEX ACOUSTIC PANEL ACOUSTIC TESTS, continued



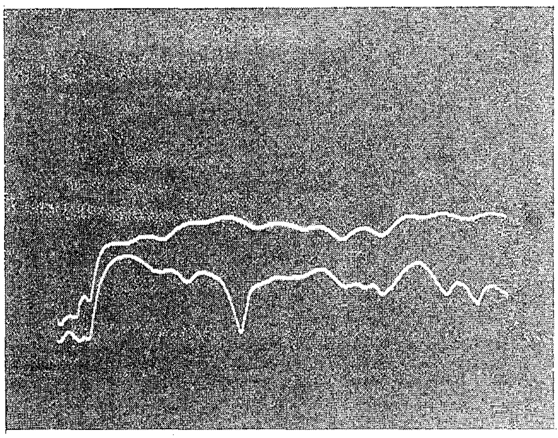
2" Indented Pattern

Here the frequency selective feature has moved down to approximately 2500 to 7500 Hz



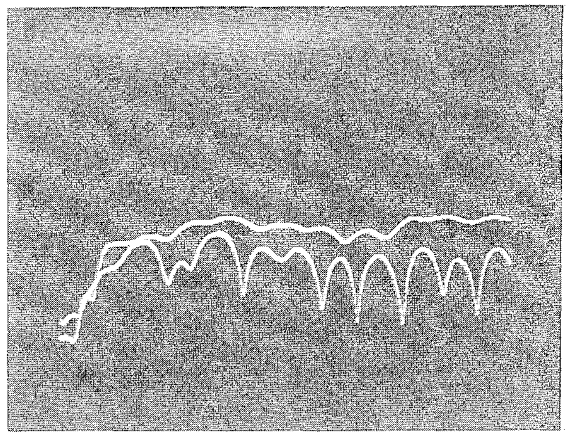
2" Wedge Pattern

Same effect. Narrower frequency range may be the effect of this pattern's better diffusion of the energy after absorption



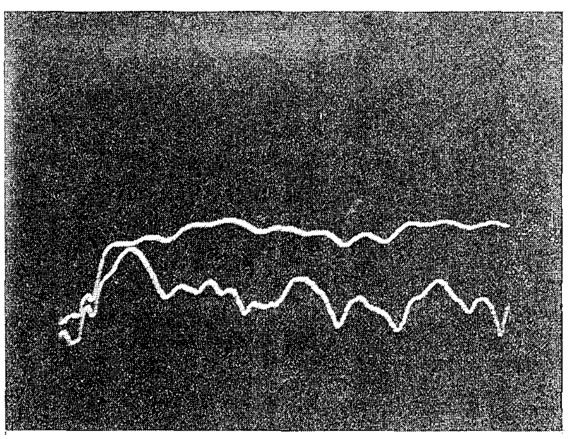
3" Wedge plus 2" Indented Pattern

The two panels were mounted on our plywood test panel, rough side out, with the 3" sample against the plywood and the 2" sample on top of the 3" sample. This resulted in a worthwhile improvement in both depth and smoothness of the reflected signal energy



Same samples as 3"+2", but with 2" sample reversed so that smooth side faced the test loudspeaker

Of interest here is the fact that the backing on these samples was a reasonable reflector, as it produced a good comb filter effect between the absorbed signal and the reflection from the backing



4" Indented + 4" Wedge + 5" separation. Rough sides out

This test indicated that these samples benefit at the lower frequencies from the usual practice of spacing

CONCLUSION

From this data we would unhesitatingly recommend this material for use in LEDE control rooms and as an exceptionally useful material to use in the construction of M_a traps.

If interested, contact Illbruck/USA, 3800 Washington Ave, N., Minneapolis, MN 55412. (612) 521-3555.

LINE LOSS PROGRAM

ED LETHERT is living proof of a powerful mathematical mind freed by the modern computer. Ed's preparation of basic computer programs for the HP 67 is thorough, documented, and beautifully researched. Here's a most useful one on line losses.

In the years prior to 1924, the telephone company used the standard cable mile (STC) as a way to express loss. If the standard cable was 19 ga copper twisted with an impedance in the neighborhood of 600Ω, then Ed's program shows that a mile (5280') of cable has a loss of 1.192 dB

<u>HP-67 Line Loss Program</u>

```
Step 1.  ENTER........... 1 = Solid Copper
                          2 = Solid Copper, Tinned
                          3 = Stranded Copper
                          4 = Stranded Copper, Tinned
         PRESS........... fa
         DISPLAY......... ohms per circular-mil-foot
Step 2.  ENTER........... Wire Size (AWG)
         PRESS........... A
         DISPLAY......... Wire Size
Step 3.  ENTER........... Circuit Length (ft)
         PRESS........... B
         DISPLAY......... Circuit Length
Step 4.  ENTER........... Load Resistance (ohms)
         PRESS........... C
         DISPLAY......... Load Resistance
Step 5.  PRESS........... D
         DISPLAY......... Circuit Loss (dB)
```

Steps A thru D are interactive and any three quantities may be entered and the fourth calculated. All entries and solutions are stored in the memory.

To calculate the actual voltage drop, after the line loss has been computed enter the source voltage and press E and the display will show the actual drop in volts.

If you are using transformers, enter the transformer insertion loss in dB and press fb. The display will show a correction factor. Now enter the sum of all of the transformers' rated watts and press 1) fc if you are using a 25 volt system, or 2) fd if you are using a 70 volt system. The display will show the actual load resistance and the results will be automatically entered into the memory.

<u>Program from an HP 67 card, printed out on an HP 97</u>

```
001  *LBLa  21 16 11
002   STOI     35 46
003    F1?  16 23 01
004   GTO4     22 04
005    SF1  16 21 01
006      1        01
007      0        00
008      .       -62
009      5        05
010      7        07
011   STO1     35 01
012      1        01
013      0        00
014      .       -62
015      7        07
016      9        09
017   STO2     35 02
018      1        01
019      0        00
020      .       -62
021      7        07
022      7        07
023   STO3     35 03
024      1        01
025      1        01
026   STO4     35 04
027      1        01
028      .       -62
029      2        02
030      6        06
031      0        00
032      9        09
033      7        07
034      7        07
035   STO5     35 05
036  *LBL4     21 04
037   RCLI     36 45
038    RTN        24
039  *LBLA     21 11
040   STOA     35 11
041    F3?  16 23 03
042    RTN        24
043   GSB2     23 02
044      2        02
045      5        05
046      x       -35
047   RCLB     36 12
048   RCLI     36 45
049      x       -35
050      2        02
051      x       -35
052    X⇄Y       -41
053      ÷       -24
054     LN        32
055   RCL5     36 05
056     LN        32
057      ÷       -24
058    CHS       -22
059      3        03
060      6        06
061      +       -55
062    PSE     16 51
063      4        04
064    X>Y?    16-34
065   GTO3     22 03
066     R↓       -31
067      2        02
068      ÷       -24
069    INT     16 34
070      2        02
071      x       -35
072   STOA     35 11
073    RTN        24
074  *LBLB     21 12
075   STOB     35 12
076    F3?  16 23 03
077    RTN        24
078   GSB2     23 02
079      2        02
080      5        05
081      x       -35
082   RCL5     36 05
083      3        03
084      6        06
085   RCLA     36 11
086      -       -45
087     Y^x       31
088      x       -35
089   RCLI     36 45
090      2        02
091      x       -35
092      ÷       -24
093   STOB     35 12
094    RTN        24
095  *LBLC     21 13
096   STOC     35 13
097    F3?  16 23 03
098    RTN        24
099   GSB1     23 01
100   RCLD     36 14
101    CHS       -22
102      2        02
103      0        00
104      ÷       -24
105   10^x     16 33
106    CHS       -22
107      1        01
108      -       -55
109      ÷       -24
110   RCLE     36 15
111      -       -45
112   STOC     35 13
113    RTN        24
114  *LBLD     21 14
115   STOD     35 14
116    F3?  16 23 03
117    RTN        24
118   GSB1     23 01
119   RCLC     36 13
120   RCLC     35 13
121   RCL6     36 06
122      +       -55
123      ÷       -24
124    LOG     16 32
125      2        02
126      0        00
127      x       -35
128    CHS       -22
129   STOD     35 14
130    RTN        24
131  *LBLE     21 15
132   GSB2     23 02
133      x       -35
134   RCLC     36 13
135   RCL6     36 06
136      +       -55
137      ÷       -24
138    RTN        24
139  *LBL1     21 01
140   RCLB     36 12
141   RCLI     36 45
142      x       -35
143      2        02
144      x       -35
145   RCL5     36 05
146      3        03
147      6        06
148   RCLA     36 11
149      -       -45
150     Y^x       31
151      2        02
152      5        05
153      x       -35
154      ÷       -24
155   STO6     35 06
156    RTN        24
157  *LBL2     21 02
158   RCLD     36 14
159    CHS       -22
160      2        02
161      0        00
162      ÷       -24
163   10^x     16 33
164    1/x        52
165      1        01
166      -       -45
167   RCLC     36 13
168      x       -35
169   STO6     35 06
170    RTN        24
171  *LBL3     21 03
172     R↓       -31
173    INT     16 34
174   STOA     35 11
175    RTN        24
176  *LBLb  21 16 12
177      1        01
178      0        00
179      ÷       -24
180   10^x     16 33
181   STO7     35 07
182    CF3  16 22 03
183    RTN        24
184  *LBLc  21 16 13
185   RCL7     36 07
186      x       -35
187      5        05
188      2        02
189      5        05
190    X⇄Y       -41
191      ÷       -24
192   STOC     35 13
193    CF3  16 22 03
194    RTN        24
195  *LBLd  21 16 14
196   RCL7     36 07
197      x       -35
198      5        05
199      0        00
200      0        00
201      0        00
202    X⇄Y       -41
203      ÷       -24
204   STOC     35 13
205    CF3  16 22 03
206    RTN        24
207    R/S        51
```

SUGGESTED LIST OF DESCRIPTORS

TABLE 1: A-WEIGHTED RECOMMENDED DESCRIPTOR LIST

	Term	Symbol
1.	A-Weighted Sound Level	L_A
2.	A-Weighted Sound Power Level	L_{WA}
3.	Maximum A-Weighted Sound Level	L_{max}
4.	Peak A-Weighted Sound Level	L_{Apk}
5.	Level Exceeded x% of the time	L_x
6.	Equivalent Sound Level	L_{eq}
7.	Equivalent Sound Level over Time (T)[1]	$L_{eq(T)}$
8.	Day Sound Level	L_d
9.	Night Sound Level	L_n
10.	Day-Night Sound Level	L_{dn}
11.	Yearly Day-Night Sound Level	$L_{dn(y)}$
12.	Sound Exposure Level	L_{SE}

(1) Unless otherwise specified, time is in hours (e.g., the hourly equivalent level is L_{eq} (1)). Time may be specified in non-quantitative terms (e.g., could be specified as $L_{eq(WASH)}$ to mean the washing cycle noise for a washing machine.)

TABLE II: RECOMMENDED DESCRIPTOR LIST

	Term	A-Weighting	Alternative[1] A-Weighting	Other Weighting[2]	Unweighted
1.	Sound (Pressure) Level[3]	L_A	L_{pA}	L_B, L_{pB}	L_p
2.	Sound Power Level	L_{WA}		L_{WB}	L_W
3.	Max. Sound Level	L_{max}	L_{Amax}	L_{Bmax}	L_{pmax}
4.	Peak Sound (Pressure) Level	L_{Apk}		L_{Bpk}	L_{pk}
5.	Level Exceeded x% of the time	L_x	L_{Ax}	L_{Bx}	L_{px}
6.	Equivalent Sound Level	L_{eq}	L_{Aeq}	L_{Beq}	L_{peq}
7.	Equivalent Sound Level Over Time (T)[4]	$L_{eq(T)}$	$L_{Aeq(T)}$	$L_{Beq(T)}$	$L_{peq(T)}$
8.	Day Sound Level	L_d	L_{Ad}	L_{Bd}	L_{pd}
9.	Night Sound Level	L_n	L_{An}	L_{Bn}	L_{pn}
10.	Day-Night Sound Level	L_{dn}	L_{Adn}	L_{Bdn}	L_{pdn}
11.	Yearly Day-Night Sound Level	$L_{dn(y)}$	$L_{Adn(Y)}$	$L_{Bdn(Y)}$	$L_{pdn(Y)}$
12.	Sound Exposure Level	L_S	L_{SA}	L_{SB}	L_{Sp}
13.	Energy Average value over (non-time domain) set of observations	$L_{eq(e)}$	$L_{Aeq(e)}$	$L_{Beq(e)}$	$L_{peq(e)}$
14.	Level exceeded x% of the total set of (non-time domain) observations	$L_{x(e)}$	$L_{Ax(e)}$	$L_{Bx(e)}$	$L_{px(e)}$
15.	Average L_x value	L_x	L_{Ax}	L_{Bx}	L_{px}

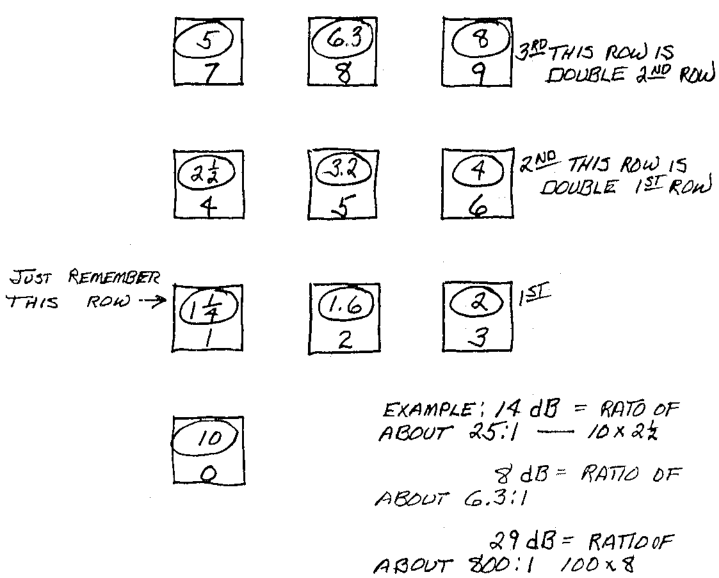
(1) "Alternative" symbols may be used to assure clarity or consistency.
(2) Only B-weighting shown. Applies also to C,D,E,..... weighting.
(3) The term "pressure" is used only for the unweighted level.
(4) Unless otherwise specified, time is in hours (e.g., the hourly equivalent level is $L_{eq(1)}$). Time may be specified in non-quantitative terms (e.g., could be specified as $L_{eq(WASH)}$ to mean the washing cycle noise for a washing machine)).

Few situations are more confused than the noise control field. The National Council of Acoustical Consultants (Syn-Aud-Con is a member) has published a suggested list of descriptors. The list looks as useful to us as any we have seen and we'll use it in SSE when we re-write it.

POWER RATIOS

TED LeTOURNEAU, Acoustic Consultants in Longview, TX, gave us "Hearing!!! How Did Jesus Talk to So Many at Once" when he was in the Orlando 78 class, which we published in the January Newsletter (we had many requests to reprint it). At the same time, he gave us the following drawing. We constructed a few problems to explain it.

ARRANGEMENT OF KEYS ON A CALCULATOR HELPS TO REMEMBER POWER RATIOS OF DECIBELS



If you are given 29 dB and asked the power ratio and all you have is a picture of a four function calculator keyboard shown in an advertisement in the evening paper you can quickly find the answer.

First you must have memorized but four numbers. These are 1¼, 1.6, 2 and 10.

If you assign these memorized numbers to the keyboard as shown in our illustration then the remainder of the numbers is merely the doubling of those numbers you already have, or, if you prefer, simply remember your 1/3 octave filter spacings: 125, 160, 200, 250, 315, 400, 500, 630, 800, and 1000, which by placing the decimal point two places to the left results in 1.25, 1.60, 2.00, 2.50, 3.15, 4.00, 5.00, 6.30, 8.00 and 10.00.

Then, 29 dB is two tens (100) plus keyboard nine or power ratio (8). Therefore, 8x100 = 800.

The power ratio is 800 to 1. 37 dB = 3 tens or 1000 x (5) (keyboard seven or power ratio) = 5000 to 1.

By reversal, a power ratio of 45 to 1 breaks down to 4.5 times ten. A power ratio of (4) on the keyboard is 6 and of (5) is seven. Therefore, it is reasonable to assume that 10 + 6.5 or 16.5 dB is near the correct answer.

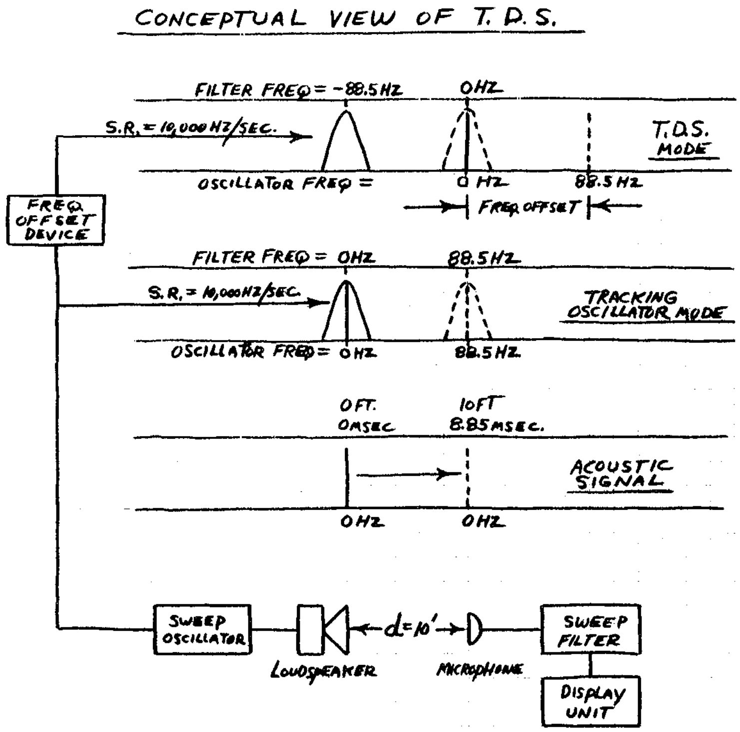
Again, 14.5 dB is a power ratio of ten times keyboard (4) which is a power ratio of 2.25 and keyboard (5) is a power ratio of 3.2.

Thus, 10 times 2.25 + $\left(\frac{3.15 - 2.25}{2}\right)$ = 2.7 x 10 ≃ a power ratio of 27 to 1.

All this can be of help when the government reaches its final energy solution.

CONCEPTUAL VIEW OF TDS

Start at the bottom of the illustration. A sweep oscillator, sweeping linearly across the frequency spectrum at 10,000 Hz every sec or 1000 Hz every 100 msec, is fed to a loudspeaker which then emits an acoustic replica of the electrical signal. Ten feet in front of the loudspeaker is a microphone attached to a bandpass filter set (usually 10 Hz bandwidth) which in *normal* operation is "locked" to the same frequency as the oscillator so that they both sweep the same frequencies at the same time - tracking oscillator mode. The output of the filter is connected to a suitable display, usually an oscilloscope.



If the sweep oscillator begins its sweep at 0 Hz and sweeps up in frequency at 10,000 Hz per second, then every 8.85 msec the oscillator will have risen in frequency another 88.5 Hz. In other words, because at this sweep rate it takes 8.85 msec to sweep over 88.5 Hz, we can say that 88.5 Hz is identical to 8.85 msec.

Looking at the acoustic signal traveling from the loudspeaker to the microphone which must of necessity follow the speed of sound in air (1130 ft/sec) then a d = 10' means that our sound has taken

$$\frac{1130'}{1 \text{ sec}} = \frac{10'}{? \text{ sec}}$$

$$? \text{ sec} = \frac{10}{1130} = .00885 \text{ secs or } 8.85 \text{ msecs}$$

to go from the loudspeaker to the microphone. This means that 8.85 msec after the start of the sweep, 0 Hz arrives at the microphone. But 8.85 msec after the start of the sweep the bandpass filter has moved from 0 Hz to 88.5 Hz and therefore the signal coming in at the microphone can't get into the display because the filter is tuned to another frequency than the one coming in at the microphone.

Looking now at the top of the illustration and observe that we have separated the tracking oscillator and filter (they still sweep at the same rate and both sweep together but the oscillator is now a set frequency offset (FO) ahead of the filter). The filter is now -88.5 Hz or 8.85 msec behind the oscillator and stays that far apart during the entire sweep. If the oscillator is at 1,000 Hz, then the filter is tuned to 1,000 - 88.5 = 911.5 Hz.

Now, when the oscillator sweeps upward in frequency and the sound travels from the loudspeaker to the microphone through the air, as 0 Hz arrives *acoustically* at the microphone while the oscillator at that same moment is at 88.5 Hz. *electrically*, the filter is simultaneously at *0 Hz electrically* and the signal is passed to the display unit through the bandpass filter.

A SHORT REVIEW OF TDS FUNDAMENTALS

TDS, Time Delay Spectrometry, is a method of measuring the spectrum (amplitude vs frequency response) of a sound source with reference to a specified time interval. A 1/3-octave RTA (real time analyzer) measures the *total* sound level vs frequency. A TDS analyzer can measure the direct sound level vs frequency whenever the specified time interval (the time window) is made short enough to discriminate against the first reflection spectrum. If, for example, the time window is made roughly 1 msec, then the analyzer will not see energy at a given frequency that is more than 1 foot longer in its arrival time than the selected energy.

If our loudspeaker emits a 1,000 Hz signal for 1 msec (1 cycle of the tone) and I have a microphone that triggers an oscilloscope upon seeing the arrival of this signal at its diaphragm and leaves the microphone output circuit on for 1 msec, then I would only receive the signal sent and no reflection could enter, say 10 msec later. This method is called pulse testing and has been available for over 40 years.

Dick Heyser conceived of and patented a vast improvement in this measurement task and called it Time Delay Spectrometry.

more.........

A SHORT REVIEW OF TDS FUNDAMENTALS, continued

Dick's idea was to sweep an oscillator rapidly enough so that a 10Hz bandwidth bandpass filter spent a very short interval of time going through any 10 Hz bandwidth. If we use a 10 Hz bandpass filter (BP = 10 Hz at its -3dB points) and sweep from 0 to 10,000 Hz (UFL-LFL) and we would like to have a bandpass sweep time (BPST) that is the time it takes the oscillator to sweep 10 Hz of 1 msec, we first calculate the sweep time (ST) for 0 to 10,000 Hz by

$$ST = \frac{BPST(UFL-LFL)}{BP}$$

and

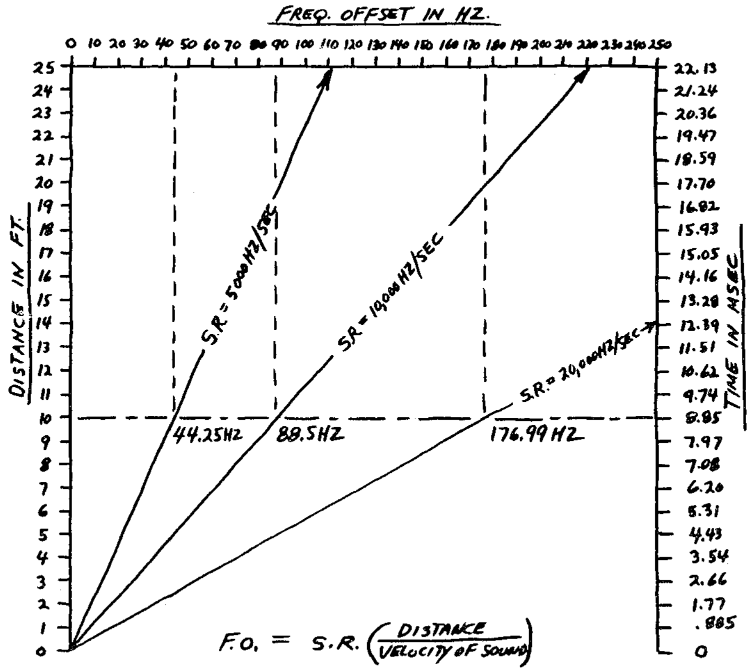
$$ST = \frac{.001(10{,}000)}{10} = 1 \text{ sec}$$

And the sweep rate SR then becomes

$$SR = \frac{UFL - LFL}{ST}$$

$$SR = \frac{10{,}000}{1} = 10{,}000 \text{ Hz/sec}$$

The relationship of these equations is graphically illustrated in Figure # 2.



Take a conventional wave analyzer, one which has an oscillator that precisely tracks its bandpass filter -- anytime the oscillator is at 1,000 Hz so is its filter etc. Feed the oscillator's output into the loudspeaker but put the microphone, for example, 10 feet away from the loudspeaker, then the sound from the loudspeaker will take 10' x .885 msec or 8.85 msec to reach the microphone. Since the filter precisely tracks the generator, it will be detuned from the frequency arriving at the microphone by

$$FO = SR\left(\frac{\text{Distance}}{\text{Vel. of sound}}\right)$$

Where FO is the frequency offset

or

$$FO = 10{,}000 \text{ Hz/sec}\left(\frac{10'}{1130'/\text{sec}}\right) = 88.5 \text{ Hz}$$

This means that the 10 Hz bandpass filter with a 10:1 shape factor has more than 50 dB of discrimination against receiving this signal.

The solution that Dick Heyser developed is to offset the tracking generator and filter. That is, let the oscillator sweep begin 88.5 Hz ahead of the filter's sweep and allow them to sweep together but locked 88.5 Hz apart. (10,000 Hz/sec x 8.85 msec = 88.5 Hz) 88.5 Hz FO is directly comparable to 8.85 msec in time - at a SR = 10,000 Hz/sec.

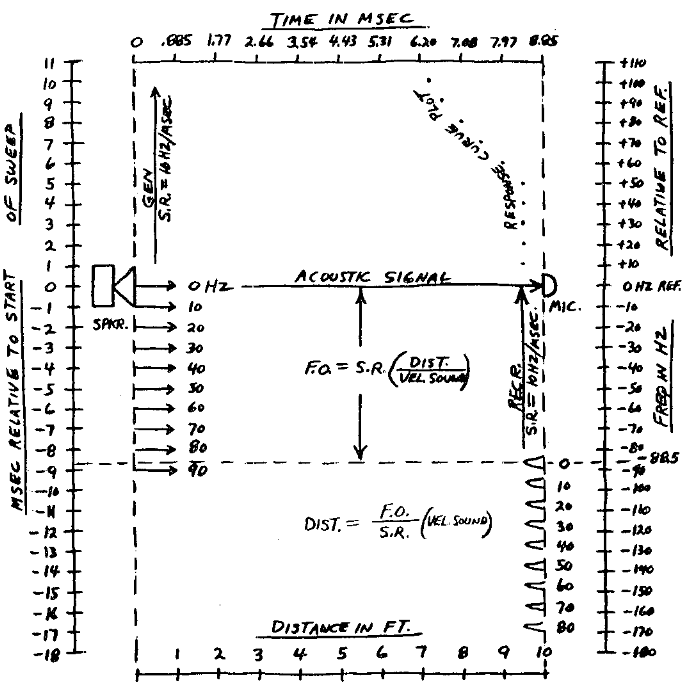


Figure # 3 illustrates visually what we have just described.

What becomes apparent is that by adjusting the FO (or time) between the oscillator and filter to be *greater* than the physical distance between the loudspeaker and microphone, reflection spectrums can be received instead of direct sound spectrums.

This simple idea (all really profound ideas are simple) is the basis of TDS. Once grasped, it opens a marvelous new world of exploration where you can still be the first man to leave your footprints!

ARTICLES ON THE RIGHT SIDE (OF THE BRAIN, THAT IS)

Dick Heyser brought to our attention an article entitled, "The Elusive Right Hemisphere of the Brain" by Eran Zaidel in the Sept-Oct. 1978 issue of Engineering and Science.

Among some of the fascinating quotes is the description of tests made on split brain patients (those who have all the cables connecting the right and left hemispheres sectioned by surgery.

> *For example, if you close the eyes of a split-brain patient and put an object in his left hand and ask him what it is, he will not be able to tell you....But, if you take the object away and mix it with other objects and ask the patient to retrieve it, still without seeing it, he will be able to do so with absolute certainty. In other words, the right hemisphere can recognize the object; it just can't tell you about it.*

And startling in its implications, if true,

> *I actually believe what is becoming increasingly accepted today, that hemispheric specialization is specified at birth.*

In the study of religious literature, perhaps the major personality changes that have occurred in Sauls becoming Pauls may indeed be fairly described as "being born again".

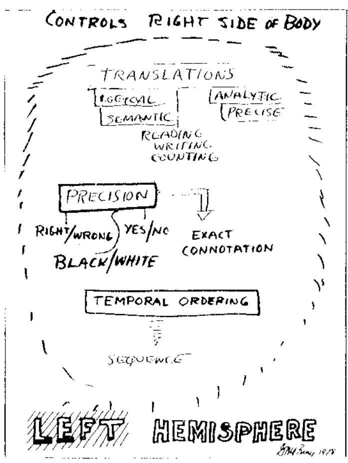
In the conclusion is the summary of right and left characteristics:

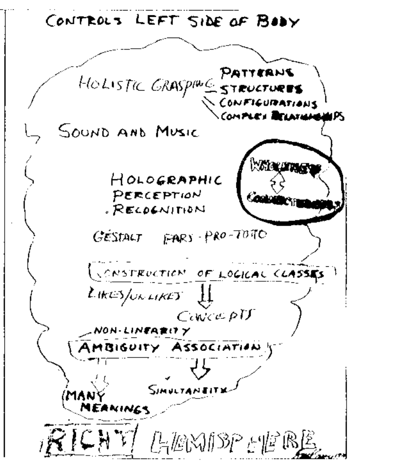
> *The left is constructive, algorithmic stepwise, and logical. It benefits from narrow examples and from trial and error; it can learn by rule.....The right hemisphere, on the other hand does not seem to learn by exposure to specific rules and examples. It does not benefit from error correction....it needs exposure to rich and associative patterns, which it tends to grasp as wholes.*

> *A team at Rutgers medical school reported that the brain's right hemisphere has a more pronounced involvement in sexual activity than the left*

It's hard to top that kind of an ending.

A second article, far less scientific but interesting because the author, George Prince, is actively engaged in attempting to boost right brain thinking in executives. This article is entitled, "Putting the Other Half of the Brain to Work", and appeared in the November 1978 issue of Training Magazine. The following illustrations appeared in the article and we felt that they were interesting to contemplate. Many Syn-Aud-Con classes have been "shoes off" meetings.





Ten ways to develop your right brain

For trainers and for learners in general, there is a goldmine in finding ways to systematically increase right-hemisphere thinking Below are 10 suggestions, outrageous and otherwise, with which you might experiment, using students, subordinates and self as subjects.

- When presenting information, have a musical background that occasionally drowns out the presentation
- Give everyone in meetings colored pens and ask them to mix their notes with doodles
- Give each participant in a meeting a lump of modeling clay.
- Give a 30-second explanation of something, and ask people to guess what you're getting at.
- At the beginning of every third meeting, set this ground rule: We can explain a point only a single time No repeating. No "in other words." If someone wants to ask a question, he does; but before getting an answer, he guesses what it will be
- In every meeting where old solutions are not working, have everyone leave his or her shoes at the door This is a signal that we will welcome confused, beginning ideas, and use ourselves to build on them rather than shoot them down
- When it is a shoes-off meeting have the person presenting the problem limit his explanation to two minutes Then ask each participant to connect the problem with an experience that is *approximately* relevant
- At every third meeting, institute the ground rule that the chairman will randomly interrupt to ask a member to describe the images going through his or her mind at that instant Other members listen to see if they can use this image to give them a beginning idea, one that does not yet work.
- One day a week, make it a rule that *no one* in the office or plant can use the word no. (The right hemisphere has no equivalent of no.) If something is not acceptable, the person must deal with it by saying, "yes, if. "
- Before a meeting when new speculative thinking is needed, have a ritual idea "dance" and light some punk so you can each read an idea in the smoke

I happen to believe that television causes atrophy of left hemisphere capabilities and that intensive reading mixed with real life visual - tactile input, not just visual - provides useful input to both hemispheres.

Novelist Bernard Malamud, in an interview in U.S. News and World Report, October 8, 1979, makes the statement that *Living is guessing what reality is.* And the provocative statement is made that, *The more I experience life, the more I become aware of illusion as primary experience.*

Shoveling horse manure is an excellent counterbalance to extended mental activity, just as riding a horse or skiing so humbles the left side as to be embarrassing. A week in a wilderness generates an appreciation of left side organization in society as a whole, while satisfying something on the right side as ancient as upright man.

ARTICLES OF INTEREST

The following quotes from the September 17 issue of Business Week Magazine ring uncomfortably true,

> *People with new ideas must be given an incentive to start companies to produce and market new products. Big business is too much the prisoner of the bottom line.*

The ideas that fuel progressive companies,

> *....require fast decision making, flexibility, and risk taking. In big companies, there is a bureaucracy designed to accomplish the opposite....What is needed is to modify the tax laws so that people who are willing to start or invest in new enterprises can make money and keep most of what they make.*

No wonder capital is being hoarded in gold and art objects by the wealthy. It's simply confiscated (called taxation) by city, county, state, and federal officials with insatiable appetites.

BOOKS OF INTEREST

The Benchmark Papers in Acoustics series has issued Volume number 10, *ARCHITECTURAL ACOUSTICS*, edited by Thomas D. Northwood. Published by Dowden Hutchinson & Ross, Inc.

This 428 page $36.00 book contains papers that are *quite difficult* to obtain. The volume is a splendid companion to the *AES Anthology on Sound Reinforcement*. Divided into four main sections: Room Acoustics--Subjective Aspects, 10 papers; Room Acoustics - Techniques and Theories, 7 papers; Sound Insulation - Subjective Aspects, 7 papers; and Sound Insulation - Technique and Theory, 6 papers.

These papers include authors ranging from Sabine, McNair, Knudsen, Moir, Bolt & Doak, Beranek, Kosten, Northwood, Young, Cremer, Cavanaugh, Schroeder, Norris, Eyring, Haas, etc.

The article, "Speech Privacy in Buildings" by Cavanaugh, Farrell, Hirtle and Watters from the JASA, 1962, remains as useful today as when first presented. Their summary of the important aspects of articulation theory (prior to Peutz's work) provides a useful synopsis.

1. The intelligible part of speech energy lies roughly between 200 and 6000 Hz
2. Most of the energy of speech is in the frequency range below 800 Hz; most of the contribution to intelligibility is above 800 Hz
3. In each frequency band speech has a dynamic range of about 30 dB; the peak values lie about 12 dB above the long-time RMS levels
4. Any frequency band in the range 200 to 6000 Hz can be considered to make a contribution to intelligibility that is proportional (a) to the fraction of its 30 dB dynamic range which is greater than the masking noise (or threshold of hearing), (b) to the bandwidth, and (c) to the importance function of that band. The importance function is a maximum at about 2,000 Hz.

MacNairs excellent article first published in JASA, 1930, entitled, "Optimum Reverberation Time for Auditoriums" contains the formula

$$\text{Optimum } RT_{60} = \left(\frac{6.35}{\text{Log } V}\right)^2$$

When applied to our sample church yields

$$\text{Optimum } RT_{60} = \left(\frac{6.35}{\text{Log}(500{,}000)}\right)^2 = 1.24 \text{ secs}$$

When applied to the $\%Al_{cons}$ equation and using a $Q = 2.5$ for a live talker and an EAD = 20' for a normal voice (MacNair Presumed live talkers) further reveals

$$\%Al_{cons} = \frac{656(20)^2}{500{,}000(2.5)} = .32\%$$

which is an ideal listening situation.

Part of the modern "golden age" of audio is the gathering of data into easily accessible form at reasonable prices. (Try paying the xerox fees for these papers separately.)

******

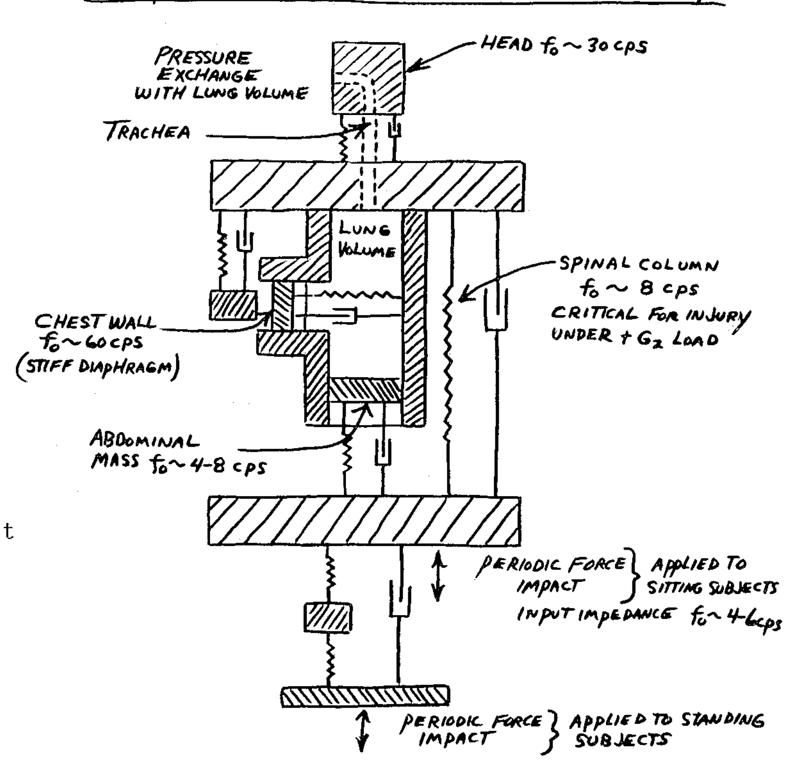
INFRASOUND AND LOW FREQUENCY VIBRATION edited by W. Tempest, Academic Press, 1976.

This 364 page $30 book is exhaustive. (The book was actually ordered by mistake but we found so much interesting material we kept it.) It covers environmental infrasound: its occurrence in measurement, infrasound in transportation, thresholds of hearing and loudness for very low frequencies, physiological and psychological effects of infrasound at moderate intensities, effects of intense infrasound on man, effects of sound on the vestibular system, subjective effects of vibration, the occurrence and its effects on performance of low frequency vibration, vibration and visual acuity, motion sickness and associated phenomena plus an appendix of the international standards in the vibration field along with a useable index.

Replete with excellent illustrations, (see sample) a detailed chart on page 212 of this volume reveals for a standing subject bowel bladder pressure is most disturbed by a frequency of from 10 to 27 Hz. Buttocks and thighs at 2-8 Hz, and chest at 2-12 Hz.

An even more explicit book than the *Shock and Vibration Handbook* by Harris and Crede, we believe it to be a most useful reference for any researcher into sound effects and other effects for theater, discos, etc.

LUMPED PARAMETER MODEL OF HUMAN BODY



BOOKS OF INTEREST

The Benchmark Papers in Acoustics, published by Dowden, Hutchinson and Ross, Inc., of Stroudsburg, PA, distributed by Halstead Press (a division of Wiley). The editor of the series is R. Bruce Lindsay.

From this series we have purchased *ACOUSTICS, HISTORICAL AND PHILOSOPHICAL DEVELOPMENT; ARCHITECTURAL ACOUSTICS;* and now, *SPEECH INTELLIGIBILITY AND SPEAKER RECOGNITION.*

This latest volume edited by Mones E. Hawley is a gold mine of early papers and includes such titles as: Calculation of the Articulation of a Telephone Circuit from the Circuit Constants, written in 1930, and Cerebral Dominance and the Perception of Verbal Stimuli by D. Kimura in 1961.

A 1957 paper by J.H. Janssen, A Method for the Calculation of the Speech Intelligibility Under Conditions of Reverberation and Noise was a precursor of Peutz's later work.

One fascinating paper by L. Lichtwitz, written in 1889, is entitled, On the Application of the New Edison Phonograph to General Hearing Measurement.

These titles hint at the diversity available in this volume. Reading these papers brings an appreciation of the value of VMA Peutz's work in this field and this book's major flaw is its failure to do no more than list Peutz in the bibliography preceding Part V. Nevertheless, it takes but a single reading to see that Peutz's equations have efficiently combined all pertinent factors far better than any predecessor. Since Syn-Aud-Con graduates already have Peutz's work in SSE and several Tech Topics, this omission is not as devastating as it is for non-Syn-Aud-Con graduates who acquire this volume.

A very short paper (one page) by Colin Cherry and Roger Wiley in 1967 in Nature contains,

> *This experiment stresses the vital importance of the temporal patterning of speech to perception.... This general result was expected on the hypothesis that speech consists essentially of accurate time patterning of sounds.*

The area of speech articulation research yet to be quantified lies here and at long last, ETC measurements allow the role of the acoustic environment to be examined in detail with regard to potential interferences between a talker and the acoustic return signals from surfaces in the environment.

******

CYBERNETICS - OR CONTROL AND COMMUNICATION IN THE ANIMAL AND THE MACHINE by Norbert Wiener, second edition, eighth printing, The MIT Press, Cambridge, MA.

> *It appears impossible for anyone seriously interested in our civilization to ignore this book,* John B. Thurston, The Saturday Review of Literature.

The above statement probably expresses the reveiwer's response to being overwhelmed by an author who can write:

> *Is it possible for a class to be infinite and yet essentially different in multiplicity from another infinite class, such as that of the positive integers? This problem was solved toward the end of the last century by Georg Cantor, and the answer is 'yes'.*

On page IX of the Preface, Weiner has , as an aside, a cognizant explanation of $ae^{i\omega t}$ as invariant under translation.

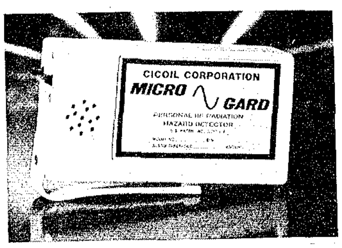
Students of Heyser, however, will find this little volume a goldmine of major proportions. Gibbs and Lebesque and their work are discussed in the Chapter on Groups and Statistical Mechanics. Mapping from one viewpoint to another viewpoint is an essential part of this book.

I'm not suggesting that the mere possession of this volume will result in profound understanding of the subject, but it will open your mind to the possible usefulness of so-called higher mathematics of one generation to the about-to-be-everyday problems of the next generation. Reading this book provides a perspective on how many enthusiasts today are rushing in with their home computers into areas where only academic angels used to tread, and then only with caution.

Instant warning of microwave exposure

Do you work near high-powered sources of microwave energy? You can now carry your own pocket-size detector to warn you when you are being exposed to radiation hazard. The battery-powered unit, small enough to be worn on a belt or carried in a pocket, features an alarm that sounds when an adjustable preset threshold level of radiation is exceeded. Based on a concept developed and patented (under NASA auspices) by the California Institute Research Foundation of the California Institute of Technology's Jet Propulsion Laboratories, Pasadena, Calif., the detector is being marketed by the Cicoil Corp.

The Micro Gard-100 weighs just 4 ounces (112 grams), including a 9-volt transistor-radio battery with a 500-hour continuous-use life, and is the size of a 100-mm pack of cigarettes. The unit's frequency range is 0.5-13 GHz, its sensitivity threshold can be calibrated to detect radia-



tion levels from 0.5 to 10 mW/cm² for alarm activation (normal calibration is at 2.0 mW/cm²), and it emits a continuous audible 1300-Hz tone of +70 dB at a distance of 20 cm as long as potentially hazardous radiation (greater than that set on the unit) remains.

The price of the Micro Gard-100 is $34.95; availability is within 30 days.

For details: Cicoil Corp., 20945 Plummer St., Chatsworth, Calif. 91311.

CLASSIFIED

FOR SALE:
UREI Sonipulse and Acoustilog Reverberation Timer.
Mike Hoover, 703-938-5115

. . SALE:
General Radio 1933-9714 (GR 1933 precision sound level meter and analyzer with built in A,B,C, and flat weightings as well as octave band filters. Complete with 1/2" and 1" calibrated microphones and GR 1562-A sound level calibrator, all in carrying case.) Like new condition. $1250.
Bob Runstein, 617-254-9200

FOR SALE:
B&K Model 1022 Beat Frequency Oscillator
B&K Model 2305 Graphic Level Recorder
$2500, or best offer
Doug Smith (evenings) 312-869-5541

FOR SALE:
Altec/Hewlett Packard 8050A Real Time Analyzer
Price??
Don Rosenberger, Ro-Cel Electronics, 731 Voice of the Theatre Building, Route 8 North, Pittsburgh, PA 412-781-2326

FOR SALE:
Crown RTA-2 with cases $1,950.
Communications Co. ARA 412RM $750.
GENRAD 1982-9720 Precision Sound Level Meter with Acoustic Calibrator $1,750.
Acoustilog 323 Reverberation Timer with options: 05, and 06 $750.
HP 8058 meter and mic $239.
Troy A. Taylor, 3350 Victor Court, Santa Clara, CA 95051. 408-968-7681 or 415-567-0575

WANTED:
HP-45 calculators in good condition
Chris Hood, 5 Harrison St., Crafton, PA 15205. (no calls please)

TED:
Project engineer for sound contractor whose work is mostly international
Contact: Andrew Ferreghy, Acromedia, 5664 Selmaraine Dr., Culver City, CA 90230. 213-390-4407

WANTED:
Sales engineer for a company on the move.
Technician-installer. Work in everything from BGM to Arenas. I.B.E.W. pay is good.
Bruce A. Thayer, WMT Stations, Box 2147, Cedar Rapids, Iowa 52406. 319-395-6000

WANTED:
Salesperson needed, experienced in pro sound, pro recording and disco sales.
Walter Horsting, WAH Sound, 1115 R Street, Sacramento, CA 95814. 916-444-5491

WANTED:
Stanal Sound is seeking an aggressive and personable salesman for their commercial sound division. Willingness to relocate necessary. Please send resume with experience, training and salary requirements
Steve Youngson, Stanal Sound, Ltd., P O Box 815, Kearney, NE 68847

WANTED:
Senior Production Mixer. Duties: Senior Staff Member for location sync-sound recording of audio tracks for motion picture and television productions. Responsible for researching project requirements, recommending techniques, assigning sound crews, and, in many cases, accomplishing original recordings on location and sound stages. Extensive travel to various locations world wide (50%). Must be familar with NAGRA 4.2 tape recorder, various microphones, booms, teleprompter, RCA magnetic film channels and various mixer boards. Minimum of six years post high school education and/or industry experience consisting of three years general background in studio recording, television broadcasting, sound reinforcement, motion picture sound and threatre. Minimum of three years specializing in audio production/post production announcing and/or teaching. College a plus - Road Work a plus. $20,611 per year.
Robert W. Houts, Chief, Sound Division
HQ Aerospace Audiovisual Services
'orton AFB, San Bernardino, CA 92409

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SYN-AUD-CON SPONSORS

Syn-Aud-Con receives tangible support from the audio industry, and ten manufacturing firms presently help underwrite the expense of providing classes in many different cities in the United States and Canada. Such support makes it possible to offer the classes in a convenient location at reasonable prices and to provide all the materials and continuing support to the graduates of Syn-Aud-Con.

Personnel from these manufacturers receive Syn-Aud-Con training which provides still another link in the communications circuit between the ultimate user and the designer-manufacturer of audio equipment. They are "in-tune" with what a Syn-Aud-Con graduate needs.

Their presence on this list as a Syn-Aud-Con sponsor indicates their desire to work cooperatively with you in professional sound.

United Recording Electronics Industries
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Shure Brothers, Inc.
Sunn Musical Equipment Company
Crown International, Inc.
Emilar Corporation
Ivie Electronics, Inc.
David Clark Co., Inc.
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