SYNERGETIC AUDIO CONCEPTS

SYN AUD CON

newsletter

Volume 13, Number 2

Winter 1986

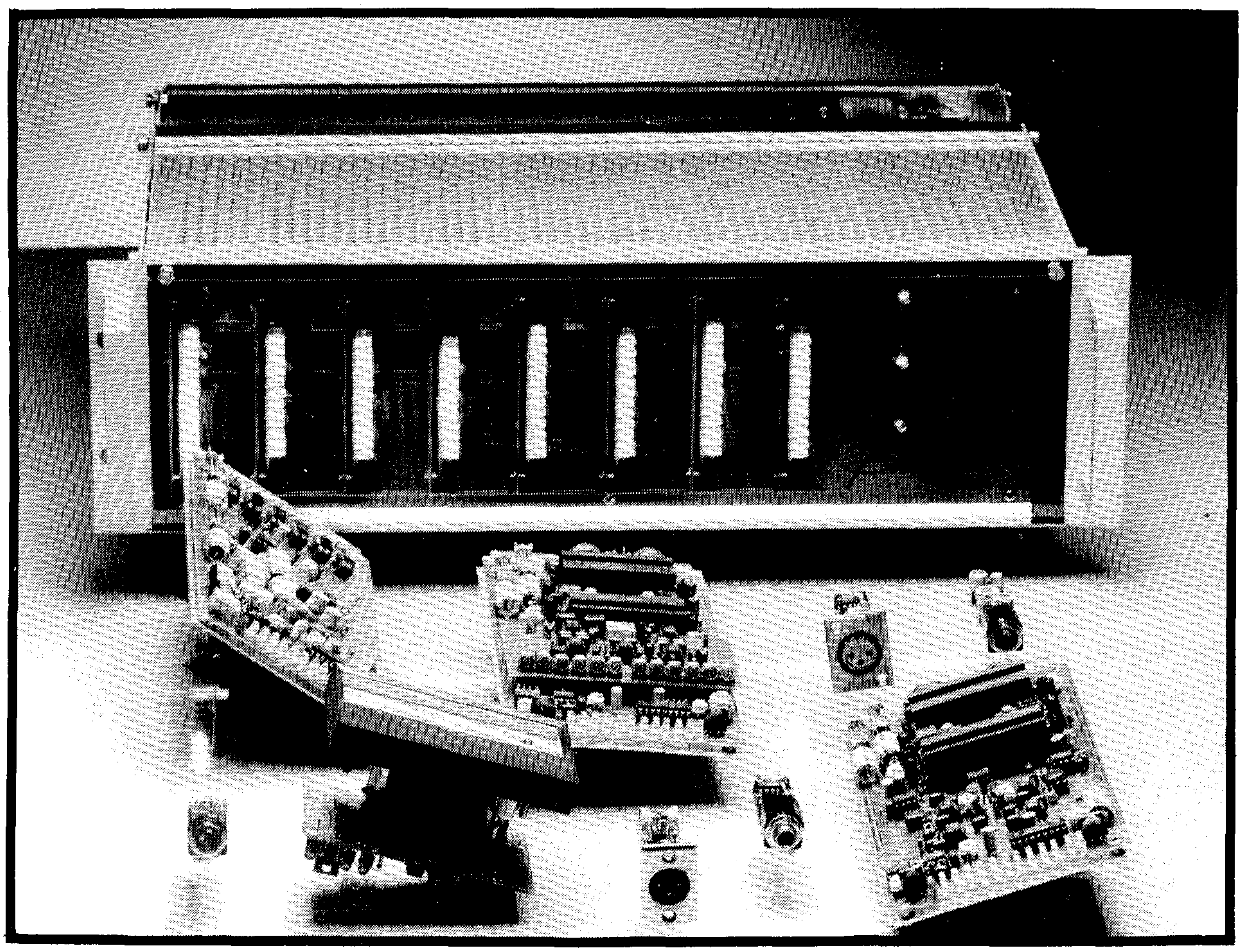
P.O. Box 669, San Juan Capistrano, CA 92693
Ph: 714-728-0245

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



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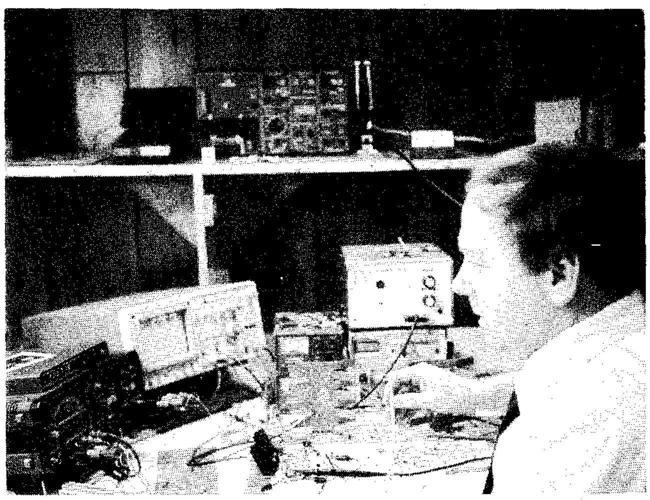
BENCHMARK - A NEW SPONSOR

Syn-Aud-Con has a new sponsor: Benchmark Media Systems, Inc., 3819 Brewerton Road, North Syracuse, NY 13212, telephone (315) 452-0400.



Headed by Allen H. Burdick, president and chief engineer, they offer mic preamp DA's, differential input devices for converting consumer electronics to balanced +4 input/output standard, and their RPM-1 meter circuits which converts existing VU meter ballistics to peak reading. Products now under development include a true stereo DA, a mic preamp card with a 1 dB noise figure, a high quality mic preamp with headphone and line outs, and a universal interface direct box with gain.

We have known Allen Burdick for a number of years and have been well aware of the quality circuit design work he has done. We are particularly pleased to have him choose to sponsor Syn-Aud-Con as he designs new state-of-the-art circuitry for his own company. We believe "Benchmark" is a very well chosen name that accurately describes the high level of performance that can be expected from these products.

Allen Burdick started in broadcast work in high school when he performed audio maintenance at a local radio station. After attending college to study chemical and electrical engineering, he joined General Electric's Visual Communications Products Department where he assisted in the development of color film chain cameras. He then went to the Caribbean where he operated a 500 kW transmitter for Trans World Radio. Upon returning to the States, Allen designed opamps with slew rates of 3,000 volts per microsecond for use in miliary applications. Later, Allen became studio technical director for CBN's

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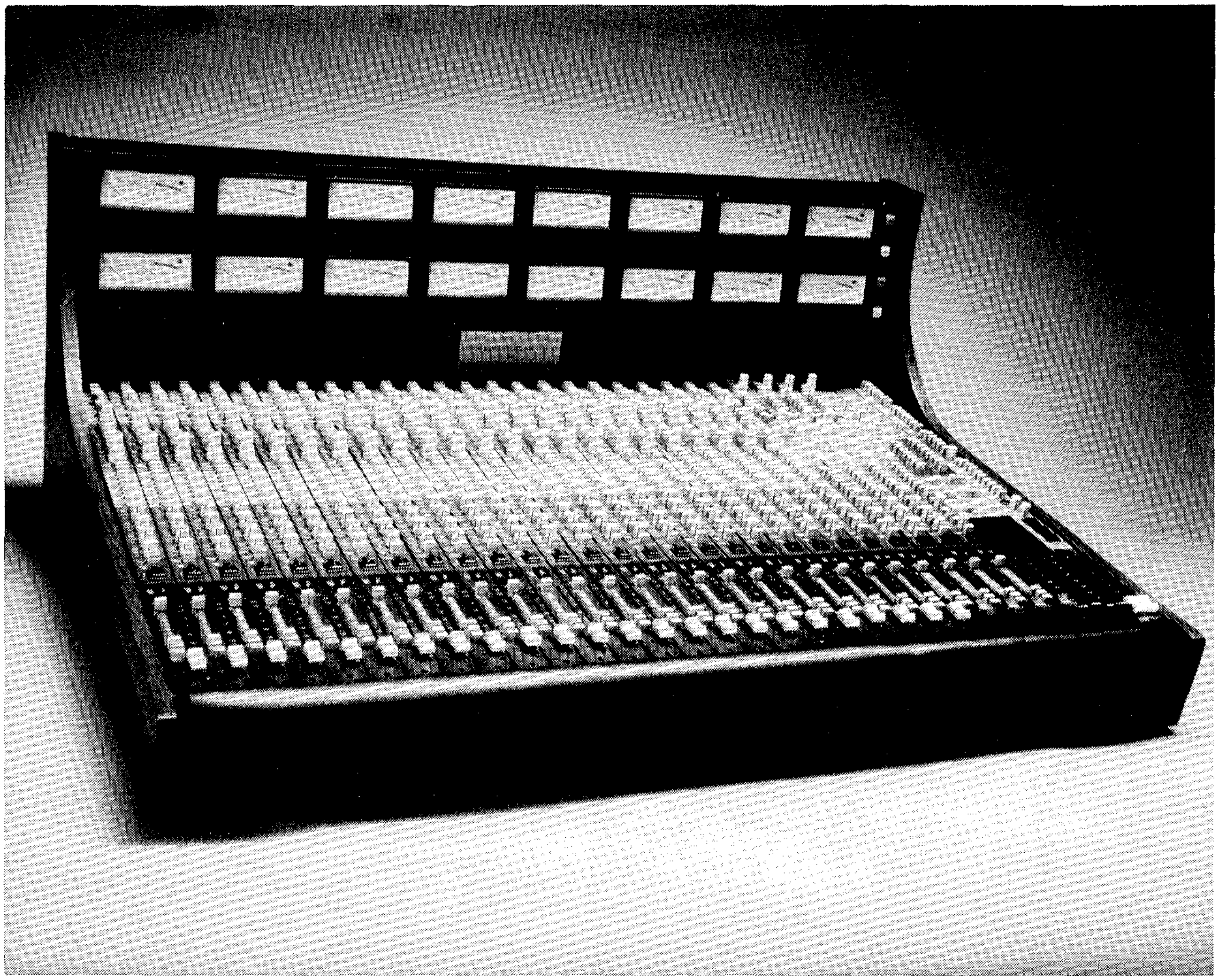
Northeast Radio Network. He then moved to Dallas to work as Systems Engineer at CBN's Dallas facility. Allen began Benchmark while in Dallas and later moved the company to Syracuse. We first met Allen in 1974 when he joined us for a Syn-Aud-Con class.

Key to the operation is David May who joined the company in 1985 in an organizational and marketing capacity. David is a many-time Syn-Aud-Con grad but he really learned his basics working for Phil Clark at DCI for many years. David's formal education in architecture gives him a strong insight into the successful integration of acoustics and electronics.

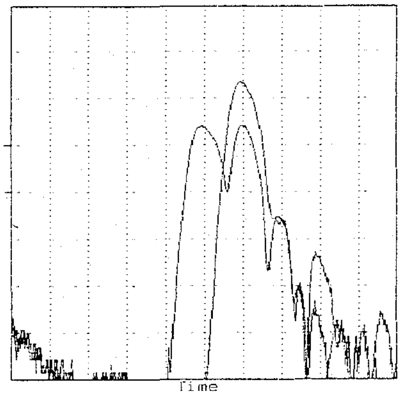


Syn-Aud-Con "grads" interested in grounding, shielding and interconnection problems should write Benchmark and obtain a copy of Allen Burdick's excellent "A Clean Audio Installation Guide." Mr. Burdick uses the term dBv, which from his usage appears to be the decibels above or below 0.775 volts. While no such standard exists, it is a common practice among circuit designers, and as long as you convert each dBv to voltage and know that it is that, it will cause no difficulty when later figuring true levels. This paper is so well written and so useful that we were glad to adapt to his use of this term.

One of the main reasons for Syn-Aud-Con sponsorship is to communicate with the market place and to provide it with the tools that it needs. Syn-Aud-Con grads know that our sponsors are especially receptive to input. Get to know Allen Burdick and David May at Benchmark Media Systems.

ELECTRONIC SIGNAL ALIGNMENT OF ALIKE DEVICES

The energy time curve (ETC) display shown here (Figure 1a) is of two measurements overlaid on the screen. The first curve shows the energy arrival over time for two alike devices that are 0.170 milliseconds (2.3") apart. The second curve shows the result of dialing in on the Sunn ADS 0.170 msec delay to the loudspeaker that originally arrived first. Note particularly the 6dB increase that occurs for coherent addition of the two signals once they are aligned. Using the Heyser disc, as we did here, provides remarkable resolution for observing such phenomenon.



VERTICAL:
6dB/div with base of display at 21.0dB

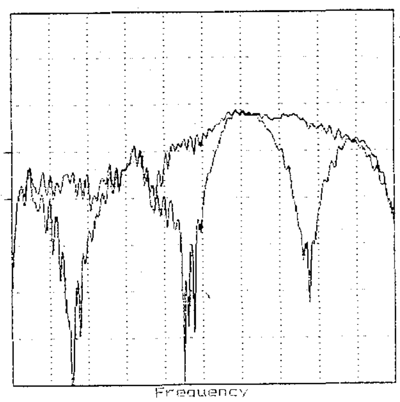
HORIZONTAL:
3000 microseconds or 3.41273 Feet to 4580 microseconds or 5.2099 Feet

Sweep rate:
5009.55 Hz/Sec

Sweep range:
99.58Hz to 31669.70Hz

FIGURE 1a.

The energy frequency curve (EFC) display (Figure 1b) is again two measurements overlaid on the screen. The first curve shows the "comb filtering" produced by two alike devices being 0.170 msec out of alignment. The second curve is the direct result of having "dialed in" the correct signal delay on the loudspeaker that had originally arrived first at the listening position.



VERTICAL:
6dB/div with base of display at 27.0dB

HORIZONTAL:
Auto 0.00Hz to 19998.10Hz

RESOLUTION:
4.5416E+00 Feet & 250.48Hz

SWEEP RATE & BANDWIDTH:
5009.55Hz/Sec & 20.00Hz

FIGURE 1b.

Several factors are of interest in these measurements. One is that the corrections were simply a matter of reading the signal delay from the TEF screen by placing the cursor on the first energy arrival and making that time interval our relative zero time. Then by moving the cursor to the center of the second energy arrival, we were able to read directly from the screen that the two devices were exactly 0.170 msec apart. We then "dialed in" that value into the Sunn delay unit and had our corrected curves.

A second factor that has come forward from doing this kind of signal alignment many times with many different devices is how wide a coverage angle the alignment remains valid for **if** the two devices are alike and individually without signal smear in their own right.

A third factor is how genuinely frustrating it can be to attempt to signal align differing devices even though they cover the same frequency range, i.e., a sectored horn and a multicell.

Experience has taught us that when utilizing superior quality loudspeaker systems, such as, the Pataxials in groups in order to obtain coverage over a large area, that superb results can be achieved with digital signal delay devices.

The only serious effects we have observed in the misalignment of high frequency units with low frequency units at the crossover region occur when the misalignment distance accidentally happens to produce a comb filter notch exactly at the same frequency as the crossover's design frequency. Some misalignments to avoid for commonly used crossover frequencies are:

Frequency	Misalignment to Avoid
500 Hz	1.13'
800 Hz	0.70'
1200 Hz	0.47'

The distance D of misalignment to avoid is found by

$$D = 0.5 \frac{c}{f}$$

where: c is the velocity of sound in feet per second or meters per second

f is the crossover frequency in Hz

D is the misalignment distance to avoid in feet or in meters.

A further caution is to avoid the other higher frequency "null frequency intervals" (NFI) that occur every multiple of

$$f = 0.5\ {}^{c}/_{d}$$

For example, f+f+f or in the case of a 500 Hz crossover the intervals become 500 Hz, 1000 Hz, 1500 Hz, etc. Looked at as a distance, we would then avoid misalignments of 1.13', 2.26', 3.39', and so on.

A BIT OF WISDOM

A highly sought after business consultant, Rosabeth Moss Kanter, is quoted in U.S. News & World Report, Dec 2, 1985, as saying on the subject of change, "How do most corporations handle change? They set up a committee." But, she adds, "You don't make changes by setting up committees. You have to have a leader with a vision."

HOUSING LOUDSPEAKERS

Syn-Aud-Con is always thrilled to see practical applications of ideas put forth in the Newsletters and Tech Topics. Michael St. Peter, State Supervisor at the Saginaw Civic Center in Saginaw, MI, sent us photographs of his theater system treated with Sonex, spring mountings, and rubber isolators.

His letter to us says it all. "We did notice a greater low-end response plus greater clarity in the mids and highs."

We salute his very creative use of Bose components.

SAGINAW CIVIC CENTER

Syn Aud Con
P.O. Box 669
San Juan Capistrano, CA 92693

Dear Don & Carolyn:

Thought I would drop you a line and some pictures of our cabinets we built for the Theater Sound System. I got the idea from one of last years Newsletters on Isolation Cabinets.

Each cabinet consists of four Bose 802's and two Bose 302 Bass Cabinets. They are mounted in seperate cells and are isolated by foam, $\frac{1}{2}$" rubber and springs. Our stage extends over the pit, so many times the performer is in front of the speakers. We added Sonex covered doors to the sides of the cabines which eliminated any feedback problem.

The only test equipment we have at the Civic Center is an RTA, but we did notice a greater low end response plus greater clarity in the mids and highs.

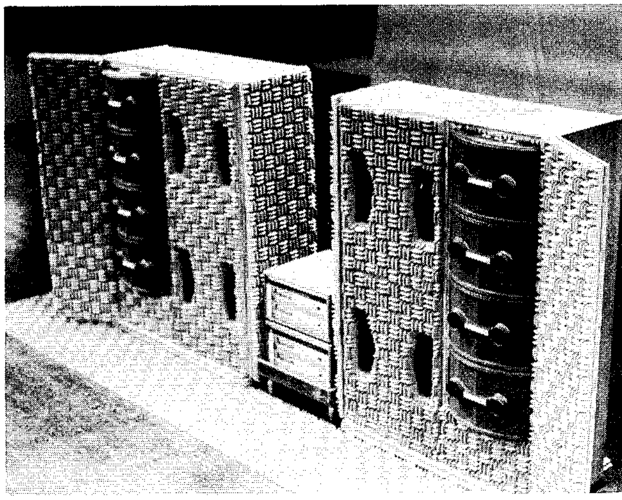
I would like to thank you again for the information I have received from your seminars.

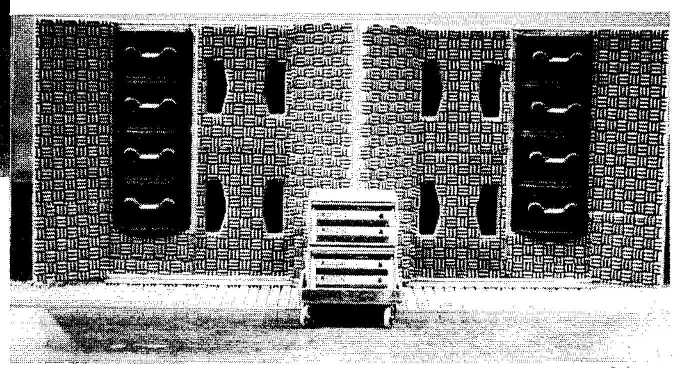
I will see you again next fall in Chicago.

Sincerely,

Michael St. Peter

Michael St.Peter,
Stage Supervisor,
Saginaw Civic Center





Michael St. Peter did a beautiful planning job. The doors can be locked as can the amplifiers. Everything is on rollers. Note that Mike has Sonex on the floor in front of the speakers to reduce floor reflection.

FOURIER DEFINITIONS

1. Fourier series

Requires a periodic waveform

2. Fourier Integral

Allows the analysis of non periodic waveforms

3. Discrete Fourier Transform

Requires N^2 operations where N equals 2^K (K = number of bits)

4. Fast Fourier Transform

Required $N \log_2 N$ operations.

For example:

A 12 bit discrete Fourier Transform requires $N = 2^{12}$ and $N^2 = 16{,}772{,}216$ operations

A 12 bit Fast Fourier Transform requires $N \log_2 N = 49{,}152$ operations

An advantage for the FFT of 341 to 1.

THE INFLUENCE OF A GOOD TEACHER

A great teacher can have a major influence in shaping a young person's career. While he was in high school, Chris Kreitz, Jan's son, used to sit in during our Workshops held at Rancho Carrillo.

Chris admired Dr. Patronis. When he joined the Navy and was chosen for University study, he selected Georgia Tech and physics. He has the "mean old doctor" (as Dr. Patronis likes to call himself) for several classes. He is now a Junior.

This picture was taken during our Atlanta class in 1985.

"CONCERT HALL ACOUSTICS"

Manfred R. Schroeder's foreword to Yoichi Ando's new book, **Concert Hall Acoustics**, contains a remarkable summary of his and his students', associates', and colleagues' work in the measurement and evaluation of existing concert halls. The foreword, from a man yet to design his first hall, is marred by criticism of those who have explored the frontiers of acoustics and tried new ideas in steel and concrete. Mr. Schroeder's stature and his accomplishments are blurred by this slip of the tongue.

The book is full of valuable data. It is not, in our opinion, a sufficient guide to insure an acceptable concert hall unless one is ready to believe that only correct reverberation and high interaural dissimilarity are all that is required. Professor Ando has presented his data in a straight forward useful manner and his discussions of interaural cross correlation coefficients, IACC, and the dip in bass response caused by incorrect seat spacing in a hall without a sufficient "rake" to the seating floor are models of measurement and analysis. Preferred delay times, preferred amplitude, and preferred direction of single reflections and of second reflections are presented.

There is no work on or apparent consideration of the density of the sound fields generated in concert halls and yet TEF measurements have suggested that this parameter varies dramatically in halls other than the classic "shoebox" design.

An interesting statement is found on page 46, "By the use of several comb filters connected in parallel...a highly irregular frequency response is produced similar to that in concert halls."

These 150 pages cost $41.50 and are worth every cent, so long as you don't get carried away with its claims, but rather adopt the tools it offers and integrate them with all the other tools just peeking over the acoustic horizon. (Publisher: Springer-Verlay, New York. $41.50)

DIRECT SOUND

One definition of direct sound is "that energy delivered by the sound source to the observer that has travelled directly from the sound source to the observer and has encountered no reflecting surfaces."

Another definition can be made that says "direct sound is that energy from the sound source plus all early reflections that arrive within (take your choice) 20, 50, 80, etc., milliseconds of the first energy to arrive at the observer".

Which definition one chooses will have important bearing on the determination of the ratio of direct-to-reverberant sound. The ratio of direct-to-reverberant sound expressed as a level in decibels will directly affect such calculations as AL_{cons}.

In studying energy time curve measurements it becomes evident that in some cases each of the time intervals mentioned above could be valid depending on the size and shape of the space, speech or music program material, and the relative levels of the early reflections.

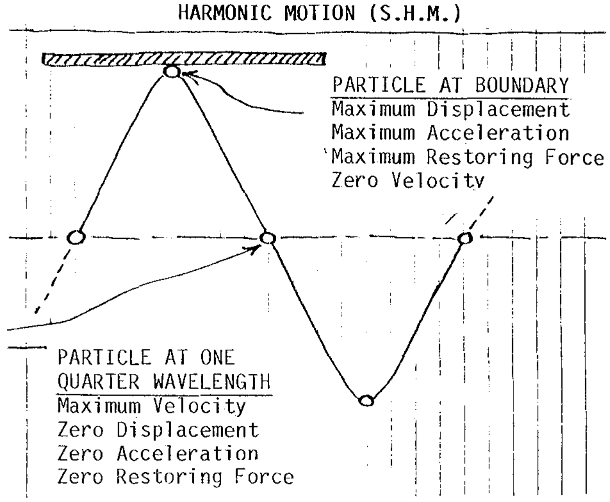
We also know the very early reflections (less than 3 milliseconds) can dramatically affect speech quality and intelligibility. So can very late, high level reflections (greater than 80 milliseconds and higher in level than the expected exponential decay at that delay).

We suggest care and an experimental attitude when approaching the vital question "What is the ratio of direct-to-reverberant sound?".

ABSORPTION

In a recent class someone asked how the mechanism of absorption worked? The answer is through friction. V.M.A. Peutz, when asked this question pointed out that when a particle encounters a boundary it is brought momentarily to a halt. Maximum acceleration occurs as it leaves the boundary and **maximum velocity occurs one quarter wavelength from the boundary.** When the absorbing fibers encounter the maximum velocity particle the maximum friction occurs (see illustration).

PARTICLE - WAVELENGTH RELATIONSHIPS IN SIMPLE HARMONIC MOTION (S.H.M.)



NEW PATRONIS SPEAKER FROM J.W. DAVIS

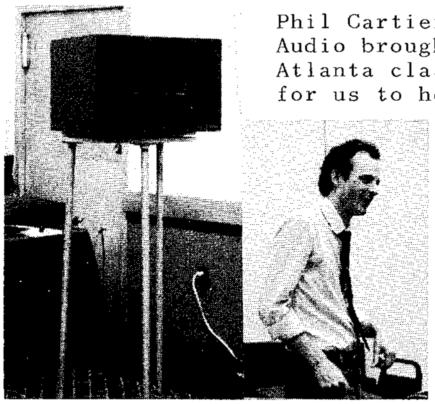
J. W. Davis and Company has just announced what we believe will turn out to be a very widely used loudspeaker, 15 1/2" high, 23 1/4" wide and 18 7/8" deep, with a weight of 56 lbs. (shipping weight), and able to handle 70 continuous watts (48.5 dBm), which means that the sensitivity figure of 95 dB at 1m/1w (EIA sensitivity 45.8 dB) allows (48.5 + 45.8) = 94 dB at 30' from full power. Dealer cost is $300!

It is fully signal aligned, wide range, easily handled package, that really makes the assembly of groups of them into large arrays totally inviting. Syn-Aud-Con plans to carry a pair of these in future classes as their range and power will allow us to demonstrate acoustic tests we're not able to do with small loudspeakers.

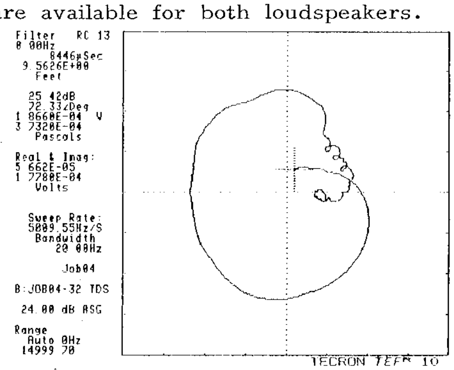
Our favorite Physics Professor talking with members of the Atlanta class

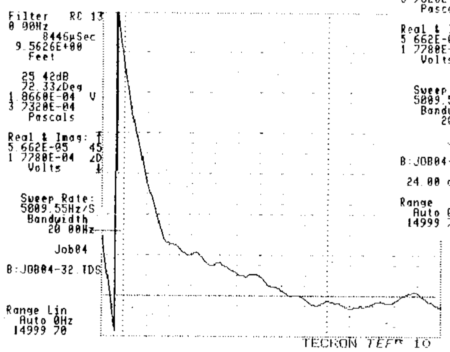
Designed by our favorite "Physics Professor" we continue to suggest that both the large Pataxial and this new one are the benchmarks we judge other offerings against.

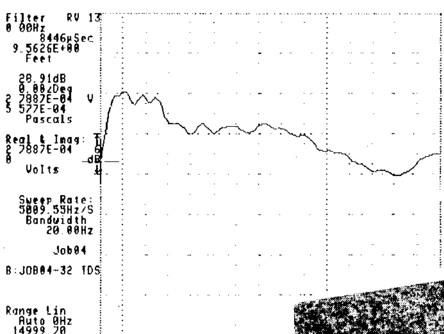
Metal grills are available for both loudspeakers.

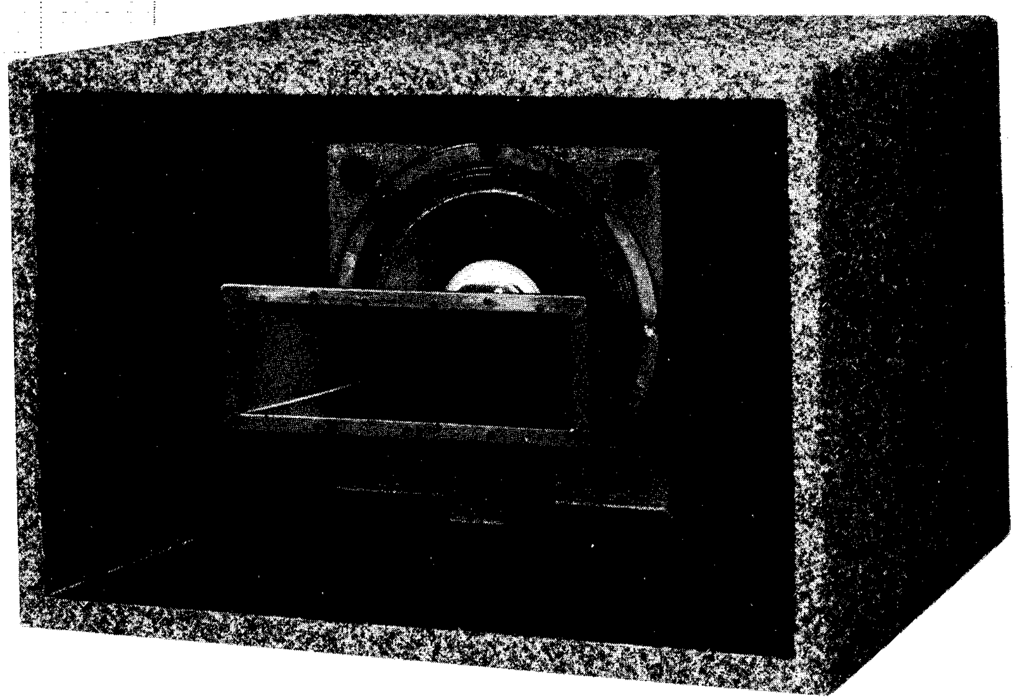


Phil Cartier of Baker Audio brought into the Atlanta class a Pataxial for us to hear and measure. We were delighted with what we heard and saw. And the price is right.









SPECIFICATIONS:

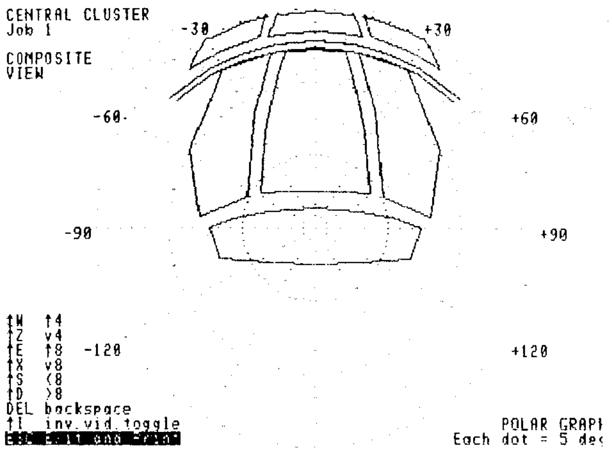
POWER HANDLING:
70 watts continuous
FREQUENCY RESPONSE:
55-18500Hz + 3dB
SENSITIVITY:
95dB/1 watt/1 meter
DIRECTIVITY:
90° horizontal x 45° vertical @ 2000 Hz
60° horizontal x 30° vertical @ 10 kHz
120° horizontal x 60° vertical @ 250 Hz

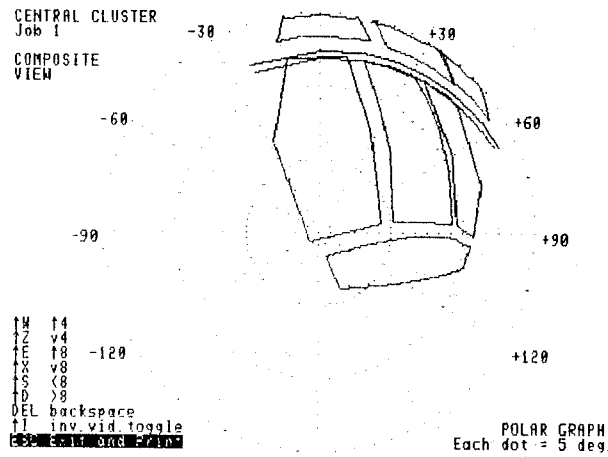
IMPEDANCE:
8 ohms minimum
(55Hz - 20 kHz)
DRIVERS:
10" woofer, 34 oz magnet
1" soft dome compression driver

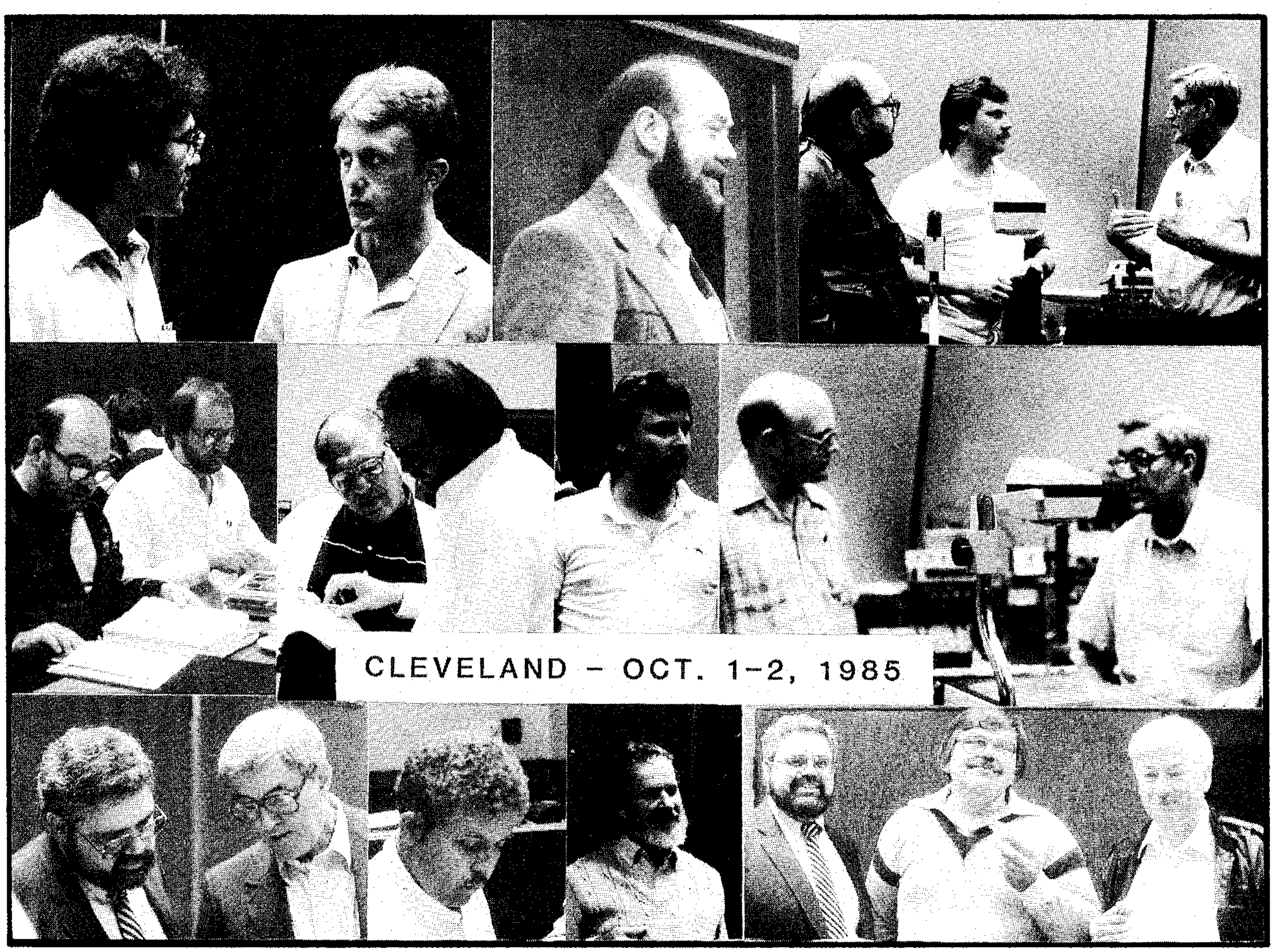
CROSSOVER:
Phase aligned,
2000 Hz
CONNECTOR
Color coded, spring loaded, cup type
(compatible with dual banana plug)
DIMENSIONS:
15½" h x 23¼" w x 18⅞" d
SHIPPING WEIGHT: 56 lbs

PHD BULLETIN

John Prohs amd David Harris are continually updating their PHD program and one of the latest features is the ability to print out the graphics. Shown here are two views of the array perspective from two different array positions. We have included with this Newsletter mailing an Update from PHD Systems & Software along with a listing of the 50 overlay patterns now available for Altec, Community/Rauland, EV and JBL.







STUDIO DESIGNER'S WORKSHOP

WHEN: May 7-9, 1986 **WHERE:** TELE-IMAGE STUDIOS
Dallas Communications Complex **FEE:** $600

STAFF: **RUSSELL BERGER,** Principal, Joiner Rose Group
Head of the design team for recording studio and broadcast facilities

DR. PETER D'ANTONIO, RPG Diffusors

DOUG JONES, Consultant to Northwestern Computer Music School

TELE-IMAGE STUDIOS: Our host for the Studio Designer's Workshop is Tele-Image, a video and audio production facility located in Texas at the Dallas Communication Complex. Within their 35,000 sq ft facility, there are three video editing suites, a screening room, a sound stage and a multi-track recording studio and a control room.

Tele-Image has taken a bold step in its committment to audio and video and there is much to be learned from the approach they have taken. We will see first hand the interrelationship and impact that acoustics has on facilitation, ergonomics, aesthetics, and functionality.

WORKSHOP CONTENT:

Russell Berger: Setting realistic performance criteria, schematic design, shapes, finishes, special acoustic systems, HVAC and mechanical noise control, environmental noise control, transmission loss of partition.

Peter D'Antonio: Control of early reflections using a reflection free zone, experimetnal reflection-free zone measurements using TDS. Creating a spatially and temporally diffuse sound field using reflection-phase grating diffusors, modal mechanics - optimizing the low frequency room response using the low frequency diffusor, design of an reflection-free zone RPG control room to satisfy LEDE criteria, measuring inter-aural cross-correlation in the control room, evaluation of existing control rooms using experimental TDS measurements, innovative future designs.

Doug Jones: How to obtain optimal stereo imagery in a control room environment, how the auditory system requires reflection free zones and extended frequency response that is free from periodic notches, however small in magnitude, in order for spatial hearing to function at its best, control room evaluation using the new LEDR tape (Listening Environment Diagnostic Recording).

LEDE and TEF technology have progressed to the place where it is possible to begin outlining preferred practices. Control room design and the thinking of control room designers has changed dramatically with the ability to measure accurately the acoustic parameters of both the rooms and the equipment in them.

Each part of the puzzle has been put into place during the last few years with increasing appreciation of the necessity to avoid early reflections and the specular region, either via absorption or geometry. The four fold role of the pressure zone, modal zone, diffusion zone and specular reflection zone from the lowest to the highest frequencies and the development of suitable tools for use in the control of each has led to remarkable control of the total sound field.

Signal alignment throughout the system, has proceeded apace with our ability to accurately measure acoustic phase. Absolute polarity, minimum phase response, easy detection of non-minimum phase response are all facets now easy to control and adjust in our electronic and acoustic systems.

The proper use of absorption, quadratic residue diffusors, resonant absorbers, specular reflectors, thermal gradients and room geometry are subject to scientific scrutiny rather than opinion.

Syn-Aud-Con Workshops are not social, political or product oriented sessions. This Studio Designer's Workshop will be devoted to exploring the outer envelope of control room technology with the best tools available under the guidance of a staff equipped to push hard on the envelope.

SYN-AUD-CON SEMINAR SCHEDULE

ANAHEIM, CA	PORTLAND, OR	VANCOUVER, B.C., CANADA	LAS VEGAS, NV
Holiday Inn	Red Lion (Jantzen Beach)	Holiday Inn - Downtown	Aladdin Hotel
January 14-15, 1986	March 26-27, 1986	April 2-3, 1986	April 28-29, 1986

ADJUSTING ACOUSTIC RESPONSE

As our acoustic measurement techniques improve, we find that we can both construct new tools and apply them in new ways. Some of these tools are:

1. Equalizers (both old and new);
2. Signal Delay Devices;
3. Frequency Dependent Signal Delay Devices;
4. More carefully selected basic components.

Let's briefly discuss what's happening in each of these categories and also what's not happening that some individuals would claim their product can do.

EQUALIZERS

An equalizer, if it is to be useful in a quality sound system, must be a **minimum** phase shift device. The parts of the system it is to equalize must also be of the minimum phase shift behavior. What this means in a very practical sense is that the amplitude and phase of the curve to be equalized has a conjugate in the equalizer that results in both uniform amplitude and uniform phase response in the total system when the equalizer is correctly applied. This is not new. Ever since the first equalizer was built by the Bell Telephone Laboratories at the turn of the century, this has been basic to their proper useage.

The new news in equalizers is IRPI's transversal equalizer which, in Syn-Aud-Con's opinion completely obsoletes all LCR equalizers of the past. Time will prove this judgment right or wrong.

SIGNAL DELAY DEVICES

Remarkably useful progress has been made in this realm. The Sunn device led the way with its full frequency response of 20 to 20,000 Hz, 90 plus dB s/n, and 10 microseconds per step (approximately 1/8"). This device **proved** several important points. The first being that 10 usecs **was not** too small an increment. The second was that alike devices can be dramatically improved in their combined response over their **entire** coverage angles. Those who have been fortunate enough to hear a demonstration of this device on a real life installation have learned how dramatic such seemingly minute adjustments can be. In many cases, the same results can be obtained by merely moving the devices back and forth, but in just as many cases the geometry required to obtain coverage necessitates electronic adjustment instead of physical adjustment. By aligning the mouths of the devices physically and the acoustic centers electronically, optimum performance is achieved.

FREQUENCY DEPENDENT SIGNAL DELAYS

Syn-Aud-Con has been inundated with crossover devices reputed to correct for signal delay between high and low frequency components. We have yet to test one that works to our satisfaction. The most common error is the failure to realize that going in and out of the delay circuit has an inherent delay of its own and that a **reference output** is required so that both high and low frequency signals are together before any further delay is applied. It is Syn-Aud-Con's firm belief that we are not likely to see a legitimate device for this purpose in the near future. Successful signal alignment of high and low frequency devices is very dependent upon the devices themselves and the manufacturer of the crossover device has no particular set of components in mind, hence has no data of what his device will do when applied to a real life load. The **only way** to look at the result of such crossovers is to observe the acoustic phase response of the loudspeakers. To see a correctly handled signal alignment, such as, the UREI 813 or the Pataxial systems built by J. W. Davis and Co. is to have a model to work towards.

MORE CAREFULLY SELECTED BASIC COMPONENTS

The most powerful economic return a TEF owner receives is the ability to screen new components and to quality control standard components used in their installations. Some devices have severe inherent smearing of their emitted signal with time. Some do not. We are increasingly desirous of working with those that do not smear. The exceptional sound system results from detailed screening of each component in it, the careful accurate setting of levels, the proper impedance relationships and precision signal alighment and equalization.

WHAT'S NOT SO

Any belief that adjustments made to the sound system electronically are going to change the acoustic environment may be safely categorized along with beliefs in a flat earth. What can be done is to minimize detrimental interaction **between** the sound system and the acoustic environment. This is usually accomplished not by electronic means, but by controlling the polar response of the loudspeakers. Equalization can play a very minor role in reducing the drive to the system at a frequency resonant with something in the room. But remember, the resonance is still there to be driven by any other source, such as, the audience.

It is well to recall that if one wanted to actively attempt to control the passive room acoustics, the device doing so would have to be non-causal, i.e., it would have to anticipate events before they actually happened.

To quote from a recent interview in the audio press, this is now possible in a room (?!?!)

> "In any room you have to deal with delays and resonances, so you have to build the inverse to those functions. Because these parts of the room behave in different ways, you have to address each section separately."

Because actual meaningful research is being carried on in the areas discussed above, many of the terms employed have appeared in print in journals, newsletters, etc. What has to be guarded against is the use of those terms without any of the actual results. We have all too frequently during the past two years found users of equipment telling us that they had this or that feature built in that, upon measurement, proved not to be so. For some reason unknown to us, more manufacturers are "faking it" than has been common in our industry in the past. The whole point in cultivating the basics is to better equip yourself to detect such impositions. TEF analysis has proven to be a powerful assist in proving or disproving claims. We eagerly await the day when we can honestly say that the majority of products offered to the audio community do what they claim to do. In the meantime, in the land of unclear thinkers, the hardheaded measurer is king.

FANTASY AND AUDIO ENGINEERING

For reasons not fully understood at this time, there are increasingly frequent cases of individuals who fantasize about objectively real phenomenon. Some of this is detectible as ignorance on the part of the fantasizer, such as is commonly encountered in articles on the decibel, wherein the writer reveals, without realizing it, that he knows absolutely nothing about the terms Gain and Loss. Some cases are not so simple and appear to the casual observer to be perpetrated by audio con men. We're sure that audio has its share of self-deluded individuals, but when we witness carefully coordinated waves of patent nonsense put forth as facts to be accepted, then several concerns surface in our thinking.

Is there really an audience gullible enough to be taken in by such tactics? Is the marketplace that manipulatable? Is there no one of impeccable authority in the audio industry with the intestinal fortitude to make a counter statement?

No one appreciates the complexity of future audio and acoustic measurements more than Syn-Aud-Con. We are therefore distressed to witness the use of very high cost test equipment in very questionable techniques touted as audio progress. Our readers have two choices. The first is to educate themselves to the best of their ability to be able to discern between the truth and false claims. The second is to throw a dart and pick the hype of your choice or the Guru that appeals to you.

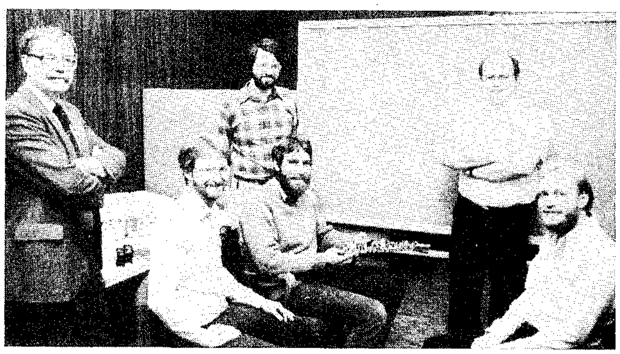
We'll repeat it one more time and you're free to take it any way you wish. The current claims being made about acoustic measurements made with FFTs are without peer group acceptance and considered, in the light of the kindest comments possible, not to be doing what is claimed for them in the popular audio press. We have no quarrel with the persons making these, to us, flagrantly false statements.

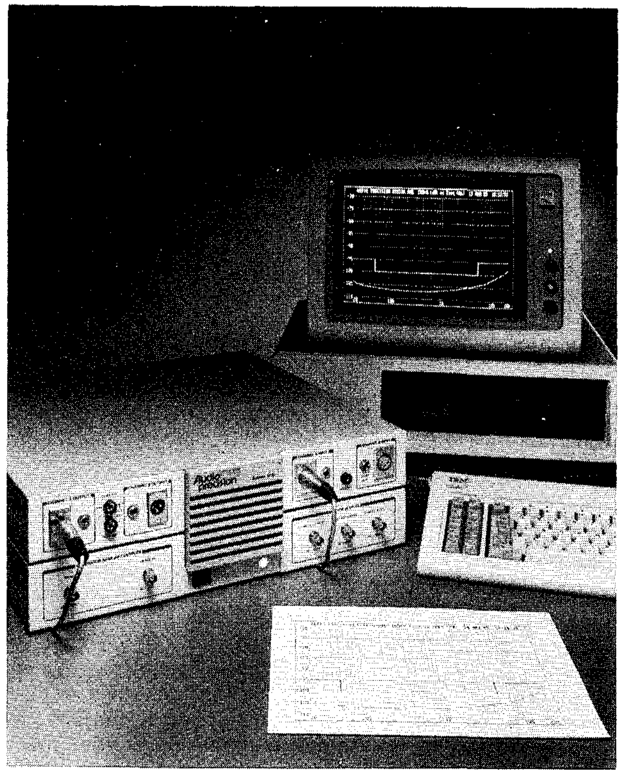
We have a very legitimate concern that audio professionals not be led expensively astray by modern versions of the confidence man. In a period where totally false articles appear even in the so-called peer group journals (unfortunately reviewers are seldom peers but politicians) we all need to be wary of what we accept as true. When the truth is established we'll be the first to help broadcast such news to everyone we can.

DR. RICHARD C. CABOT & AUDIO PRECISION

Dr. Richard C. Cabot, formerly of Tektronix and now part of the team that founded Audio Precision, P. O. Box 2209, Beaverton, OR 97075, recently sent us some preliminary data on their new measurement systems utilizing IBM-PCs. So far they are producing conservative useful devices and have resisted the FFT craze, but then Dr. Cabot and his associates are measurement professionals. One of the explanations in their literature that reflects more than the usual "smarts" reads:

> "Two methods for measuring gain or frequency response are voltage gain and transducer gain. Voltage gain techniques ignore system impedances; it compares the voltages at the input and the output of the device under test. Under the transducer gain method, generator terminal voltage is never measured. The concept is that loading effects due to device input resistance and reactance will affect the generator the same as they affect the device which normally drives the input, given a generator impedance equaling the normal drive source impedance. Loading effects then become part of the gain or frequency response measured at the terminated output of the device under test."





Dr. Cabot's measurement system allows either type of measurement.

While limited in scope at the present time, the general philosophy of Audio Precision's approach promises many benefits **if** they make the correct future choices for expanded measurement capability.

PROBLEMS AT AES?

We take no pleasure in pointing out that the AES Journal has fallen to new lows of technical creditability. A series of serious abuses detrimental to any thinking person's support of the Society is occurring much too often: Publication of a new digital standard dictated by Japanese manufacturers who as Barry Blesser has aptly described "would now be in jail if they were American manufacturers", (db Magazine Sept/Oct 1985), and publication of articles that defame long established legitimate work with measurements even the neophyte can detect are incorrect.

We fully appreciate the political nature of such enterprises and the desire to make it into a self perpetuating mutual admiration society. Perhaps that will be the final fate of the AES.

But, the audio industry badly needs a forum for the publication of accurate, truthful, useful papers on science, engineering, and technical skills worthy of our interest and study.

The AES standard on digital audio has carefully removed any serious chances of an American manufacturer participating in either consumer or recording products in the near future. To disallow competition via a premature standard is market manipulation of the worst sort and we suspect won't succeed in spite of its being sanctified by a standard.

We reluctantly remain members of the AES with the rapidly diminishing hope that some "White Knight" will ride to the rescue and flush out the permanently appointed clique. It's time the AES again becomes a society run by its members.

The thinking expressed here is not just thinking expressed by Syn-Aud-Con. There are some very serious accusations being circulated within our audio industry about the current state of affairs within the AES.

A CLARIFICATION - ALTEC

Our editorial in the last Newsletter, Vol 13, No. 1, Fall 1985, referred to audio companies destroyed by being acquired by conglomerates. In the case of Altec, the conglomerate that presided over its bankruptcy and eventual sale was LTV. It has been called to my attention that Gulton Industries, who bought Altec, has every intention of restoring it to an independent competitive status once again. Indeed they **have** restored Altec's former symbol "the maestro."

Syn-Aud-Con sincerely hopes that the symbology leads directly to full restoration of the spirit manifested by the original Altec. The United States audio industry can ill afford to lose quality companies. We'll watch with interest and encouragement to see if the intent can indeed be translated into the fact.

Simplified 1040

Latest Revision for:

1040 Federal Income Tax Form 1985

Department of the Internal Revenue Service

Your Social Security Number

Part One Income

1. How much money did you make last year? ....

2. Send it in. ....

SIGNAL TRACER

Joe Davidson of Long Communications Group in Winston Salem told us of a very useful signal tracer MODEL 77A. It can be ordered from Specialized Products Company. Joe says it really pays for itself when a wire needs to be identified.

Specialized Products Company has a very worthwhile catalog, worth the effort to obtain. Their phone is (800) 527-5018 and they have offices in Houston and Dallas.

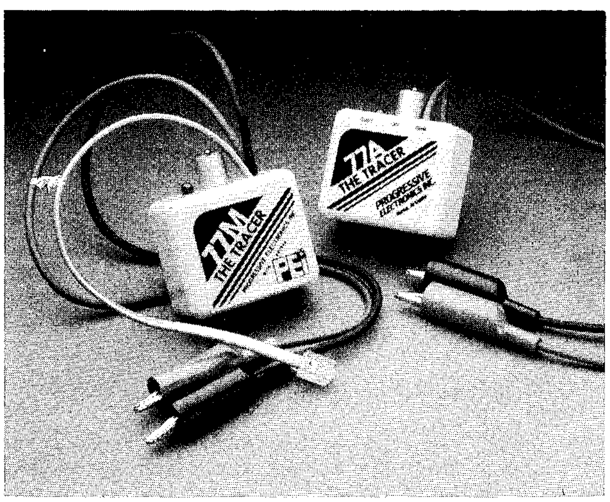
Above (Right) Joe Davidson during our New York class in 1985.

HELP! Need Publishing Software Advice

We would like to buy software for the IBM to help us lay out our Newsletter. Many of you have been computer pros for years. If you have had experience with a software you think we should try, please let us know.

We're using Word Perfect for word processing and it is very satisfactory, but we would like to add software for publishing.

Excuse us while we change over from electronic typewriter to computer. We have some material that is in old format that we will use along with new column material.



Model 77A Tracer

- Sends tone.
- Provides "talk" battery.
- Tests continuity.
- Verifies line.
- Identifies Tip and Ring.
- Indicates line condition, voltage, polarity and resistance using LED.

367S077 *77A Tracer* .... *1-5,* **$25.00 ea.,** *6 up* **$23.00 ea.**
367S079 *77C Case (either Tracer)* .... **5.00 ea.**

Model 77M Tracer

Same as 77A with additional modular connector for all test functions.

Model 77M is a continuous or alternating tone generator, continuity tester, voltage tester and resistance tester used with modular connectors in the telephone industry. It generates a distinctive audio tone (approximately 1000 Hz continuous or 1000 Hz ±30% Hi-Lo alternating) detectable with a headset or an inductive amplifier.

This Tracer is compatible with ESS, Step and Crossbar, and has an output tone that is isolated from all DC voltages.

By attaching the leads of the unit to a wire or cable all tests or indications may be detected, traced or located, either by audio means or by watching the light-emitting diode (LED). The tone and/or the continuity tests are controlled by the toggle switch.

367S082 .... *77M Tracer*
1-5, **$28.00 ea.**
6 up, **25.00 ea.**

ELECTRICAL BASICS OF SOUND SYSTEMS

When we restrict our acquisition of fundamental-principles to the needs of working with sound **systems** rather than individual components, we find that we take a different perspective than that held by component designers and service technicians.

Sound systems take the acoustic signal generated by the performer, at some point of reception, process it, and finally generate the needed acoustic **power** to supply each member of an audience with his or her share of the program material.

Component oriented individuals often are confused by the systems engineer's insistence on **power-**oriented approaches when he or she has worked successfully from a voltage oriented approach for years.

The systems engineer's insistence is based on the requirement to be able to view each component as an element in a single line block diagram and have a "gain" or "loss" assigned to each component that describes **what happens at the output of the system** when that single component is **inserted** into the system. Taking

$$20 \text{ LOG} \left(\frac{V_{out}}{V_{in}}\right)$$

unfortunately does not give the correct value except in special isolated circumstances which are not typical of modern sound system design.

MODERN SOUND SYSTEM DESIGN

Current practices follow these general trends. Microphones used in professional sound system work are normally low impedance (150-250 Ω). Mixers normally have actual input impedances of from 800 - 3000 Ω. The mixer's output impedance is normally in the 90-150 Ω region. Most intermediate active components, such as line amplifiers, equalizers, compressors and other signal processing equipment have relatively high impedance inputs and low impedance outputs of values similar to that for mixers. Power amplifiers in use today will vary in input impedance from 600 Ω for balanced line inputs to transformers to 100,000 Ω for single ended inputs. Power amplifier outputs are very low impedance, usually under 0.1 Ω and intended to be connected to loads of 4 to 16 Ω. The philosophy behind these values is the ability to "bridge" more than one device across each output as well as optimized signal-to-noise ratios.

MATCHED VS UNMATCHED CIRCUITS

Often times the component oriented engineer has encountered only the system output version of the decibel power level equation.

$$10 \text{ LOG } \frac{\text{Power in Load}}{0.001 \text{ Watt}} = \text{Level in dBm}$$

The term "dBm" stands for "the decibel level referenced to one milliwatt (i.e., 0.001w)." Note that **it is not** a voltage nor does any impedance value enter in. The reference is one milliwatt period. If a negative value appears, then the power **level** is less than one milliwatt. If the sign is positive, then the power level is greater than one milliwatt. Some typical **levels** within sound systems are:

Microphones	-45 to -70 dBm
Mixers output	0 to +30 dBm
Power amplifier	1 w = +30 dBm
	25 w = +44 dBm
	50 w = +47 dBm
	100 w = +50 dBm
	200 w = +53 dBm
	1000 w = +60 dBm

AVAILABLE INPUT POWER LEVEL (L_{AIP})

Now, if you have not encountered it before, it is not unreasonable to ask, "What do you mean when you say a microphone has a **power** output level" since microphones essentially generate a voltage across an open circuit (E_O)? Indeed, that is how they operate and how they should operate. Going back to the beginning of this discussion you will recall that what we called the "gain" or "loss" of a component was defined by what **inserting** that component did **at the output of the system.** In order to find that figure, we need a value for the input to a component that when subtracted from the output of that component accurately provides the gain or loss that its insertion into the system causes at the output of the system. The value we seek is called the **available** input power level and is the level that would be generated **if** the device were terminated in a value equal to its source impedance and could still generate the indicated voltage that was developed in the open circuit case. A measurement of a true open circuit voltage (E_O) may be treated as a source voltage(E_S)

$$E_O = E_S$$

The available input power level equation is:

$$L_{AIP} = 10\text{LOG} \left(\frac{(E_S)^2}{0.001R_S}\right) - 6.02 \text{ dB}$$

It is important to note that the L_{AIP} is in no way dependent upon the device it is to be connected to. It is also important to recognize that since the device it is to be connected to does not enter in, neither does whether that device is a match or an open circuit. The L_{AIP} is not what power level is being developed, but what power level is **available** if that same voltage were to appear across a matched load.

A SIMPLE EXAMPLE

Suppose that when your performer uses his microphone you calculate that its L_{AIP} is -60 dBm and that at the output of your system you are generating 100 watts (i.e., +50 dBm) then the **total** system gain is:

$$(+50 \text{ dBm}) - (-60 \text{ dBm}) = +110 \text{ dB}$$

Note here that the gain figure is a ratio hence in dB, but that the levels were absolute values, hence in dBm.

Continued..

BACK TO THE BEGINNING

The electrical basics we need to have thoroughly in hand therefore reduce to:

1. The open circuit voltage E_0 out of each device.
2. The actual input and output impedance of each device except for the microphone (no input Z needed) and the power amplifier where the **load** impedance is used, not the amplifier's source impedance (i.e., actual output impedance).
3. The system output voltage across the rated load.

If you have these basic parameters fully in hand, you are in a position to correctly set levels and to detect impedance anomolies that identify special cases (i.e. passive components requiring the establishment of a matched circuit).

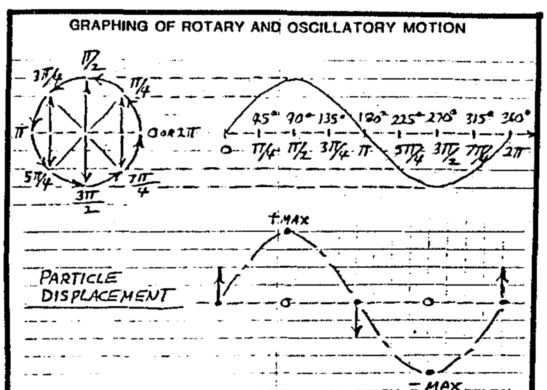
CONCLUSION

The basics outlined here are simple, based on flawless logic of over sixty years of system useage, and fundamental to the proper adjustment of any sound **system.**

PARTICLE PHYSICS

DEFINING PARTICLE POSITION, PHASE, DISPLACEMENT, VELOCITY, ACCELERATION AND RESTING FREE

Relative phase can be discussed in degrees, radians, or fractions of a wavelength. The dynamics of a particle in simple harmonic motion in a plane wave can be graphed in a simple manner. The budding acoustician should have clearly in mind the physical motion that a graph represents. Electric generators rotate, particles oscillate, water waves undulate. All can be graphed in an identical manner, but that doesn't mean they are in the same mode of motion. The illustrations here can help you visualize the different motions and their common graphical depiction.



	DISPL.	VEL.	ACCEL.	RESTORING FORCE	PHASE
	0	+MAX.	0	0	0
	+MAX.	0	-MAX.	-MAX.	$\pi/2$
	0	-MAX.	0	0	π
	-MAX.	0	+MAX.	+MAX.	$3\pi/2$
	0	+MAX.	0	0	2π
	+MAX	0	-MAX.	-MAX.	$5\pi/2$

RASTI

The world has now been blessed with a five figure "Speech Transmission Meter." RASTI stands for rapid speech transmission index and has become a pet project of Bruel & Kjaer in terms of pushing through a measurement standard (IEC draft publication 268, part 16) on its behalf. Based on the Modulation Transfer Function equations so actively written about by Houtgast and Steeneken of the Netherlands, all parties concerned have acted as if V.M.A. Peutz and his highly advanced work on the prediction and measurement of percentage of articulation loss of consonants, $\%AL_{cons}$, did not exist.

Peutz's new TEF disc for the measurement of $\%AL_{cons}$ directly is, in Syn-Aud-Con's sincere opinion, so far ahead of the RASTI system as manifested by Bruel & Kjaer as to cause concern for those who literally spend TEF money on a single use meter.

The sales literature is extensive, expensive, persuasive, and if believed, you'll be hard put to restrain yourself from rushing out and purchasing one.

A little reading of the fine print in these same documents reveals:

1. The test signal only puts out 59dB at 1m at 500 Hz and 50 dB in the 2KHz band. (In an NC -50 church, that should be fascinating.)
2. Linear transmission (total system) is assumed.
3. Pure tones in the background spectrum outside the 400 Hz and 2 KHz bands are not allowed for.
4. The RT_{60} of the room should not be strongly dependent upon frequency.

In other words, this meter should do well on an under driven system in a very quiet room with a flat reverberation curve.

If V.M.A. Peutz had never done his work, RASTI would be a real breakthrough. Since Peutz's work does exist, we can't help wondering just what's going on both in the politics of generating standards and with former leaders in the world of acoustics instrumentation.

SUNN ELECTRONICS

Sunn Electronics was one of Syn-Aud-Con's earliest Sponsors. Ably administered by Larry Lynn, they did well in their chosen field of M.I. sound. They were well know to Syn-Aud-Con grads through their exceptional ten microsecond signal delay units. Therefore it is with genuine sadness that we heard that Sunn had been purchased by Fender and that the talented personnel at Sunn has been dispersed. Sunn was owned by a mini-conglomerate and some corporate "bean counters" sold them off. We know of only one ex-Sunn person (Rex Baker from the engineering department) who has been moved to Fender, and he had worked at Fender before going to Sunn. Fender's display at NAMM exhibited selected Sunn products.

NEW TEF OWNERS

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David Klepper
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Masque Sound
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A BIT OF WISDOM

Blessed are the flexible

for they shall bend and not break

from Phil Goodman - author unknown

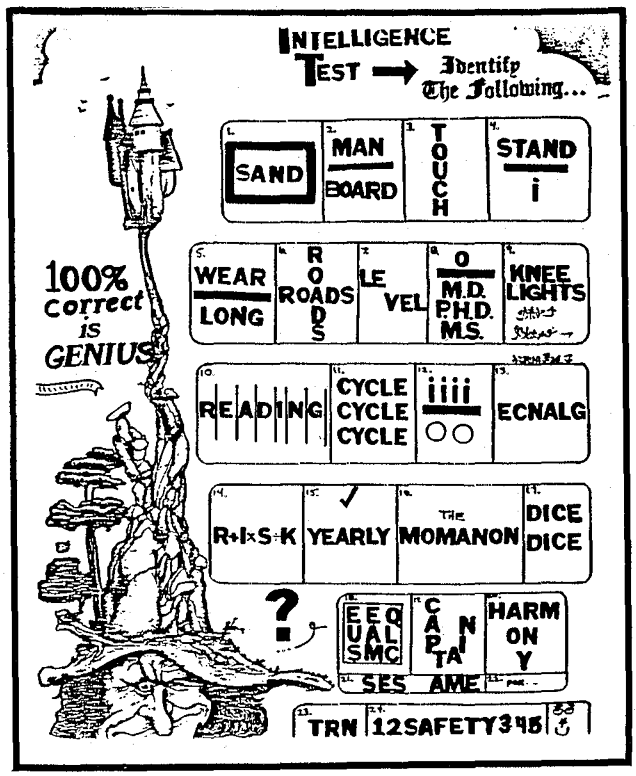
WHY TEF MEASUREMENTS

Why TEF measurements in our Newsletters when only a couple hundred TEF analyzers are out there and there are perhaps thousands of RTAs?

We could have asked a similar question in 1968 when we introduced the first RTA and the first 1/3 octave equalizer and the first computer programs for sound system design. We can look back and say how silly, how could we get along without these valuable tools? But, a real time analyzer cost $10,000 in 1968 dollars, a 1/3 octave equalizer was almost $2,000 and the first desk top HP computer/-calculator cost $5,200 (and it had less calculating capacity than an HP 41CV). Those were the tools that we owned and we waited 15 years for the these tools to become so common place that few people even think about where they came from or why we have them.

We use the TEF measurements in the Newsletter because the industry is ready to learn. Not everyone wants or needs to buy a TEF but if everyone will take the time to learn to read the measurements they can learn the valuable lessons that those with TEF machines are willing to share - like the importance of housing loudspeaker arrays, the necessity of alignment of arrays, the reasons why loudspeakers and microphones must be kept away from reflecting surfaces, why reverberation is of little significance compared to late reflections and high signal-to-noise ratios.

If anyone has a question about any of the measurements, how to read them and what they mean, don't hesitate to call or write.



NEW STUDIO MONITOR

Bob Todrank of Valley Audio in Nashville has contracted with Neil Grant of Wembley, Great Britain to market his powerful "Boxer" direct radiating, tri-amplified, studio monitoring systems. Powered by 1200 watts to 2 ohms at the low end. 485 watts to 8 ohms to the mid-section, and 250 watts to 8 ohms to the high section along with a sensitivity rating of 95 dB at 1 meter at 1 watt and a frequency response quoted to be down 3 dB at 34 Hz radiating into 2 space this unit should reach levels of 121 dB at one meter at 34 hz even when down 3 dB.

These are expensive units and come close to falling in the category of "if you have to ask the price you can't afford them." They come complete with electronics including the equalizers. To quote from the brochure:

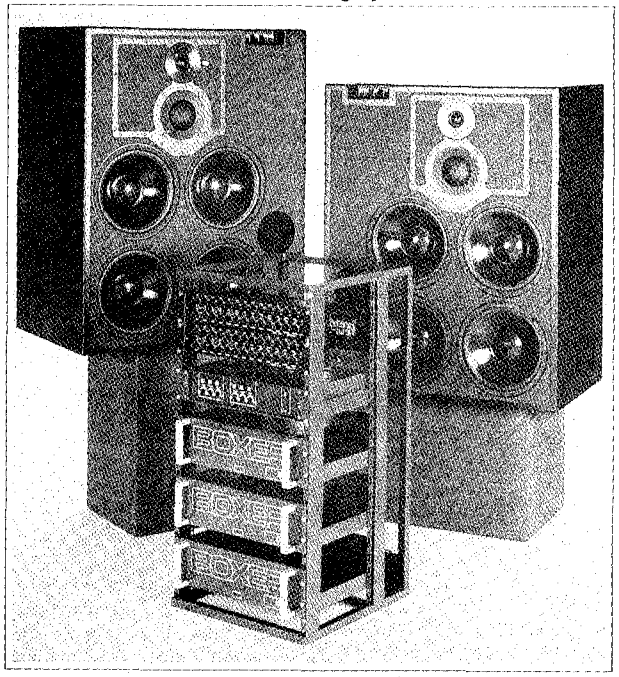
> The price of the system includes the provision for TEF commissioning anywhere within the United States or Europe at no additional cost other than travel expenses. On completion of commissioning each client is provided with a brochure containing all the TEF curves taken within his/her control room. This information is retained on floppy disc for comparison at a later date. The floppy disc remains the property of the client, and the information contained thereon is confidential to that client.

For further information, contact Bob Todrank at Valley Audio, P O Box 40743, Nashville, TN 37204, Ph 615-383-4732.

Oct. '85 PRELIMINARY

BOXER

High Power, Direct Radiating, Tri-Amplified Studio Monitoring Systems



BOXER 4 Studio Monitor System

A "PROOF" THAT TWO EQUALS ONE

Let: $x = a$

Then: $x^2 = ax$ Equals multiplied by equals remain equal.

And: $x^2 - a^2 = ax - a^2$ Equals taken from equals remain equal.

Factoring: $(x - a)(x + a) = a(x - a)$

And dividing: $\frac{(x - a)(x + a)}{(x - a)} = \frac{a(x - a)}{x - a}$ Equals divided by equals remain equal.

Canceling, we obtain: $x + a = a$

But $X = a$ so: $2a = a$

Therefore: $2 = 1$

Question: What happens on a well-designed computor if you run this problem, and why? #

TECHRON'S WORKBENCH SOFTWARE

Techron's Don Keele has programmed the TEF analyzer for use as a function generator, oscilloscope, digital voltmeter, and two types of sound level meter. Don made this a "menu driven" program and succeeded so well that we were able to use it without any other instruction than the menus themselves. We were using a preliminary version that does not have the instruction implementation yet and we had no difficulty making all the set-ups.

We have reproduced the menu and display printouts here to provide an idea of how each of these tools was programmed.

Just a few of the discs we now use regularly and would hate to be without are:

1. The Techron 2.0 version
2. The latest Heyser disc
3. The Peutz %AL_{cons} disc
4. The Peutz 3-D RT_{60} disc
5. The PHD disc
6. The workbench disc
7. The FTC disc
8. Farrel Becker's version of 2.0
9. The Log 3-D disc

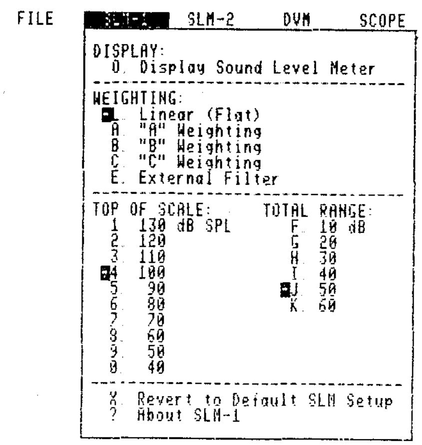


FIGURE 1b

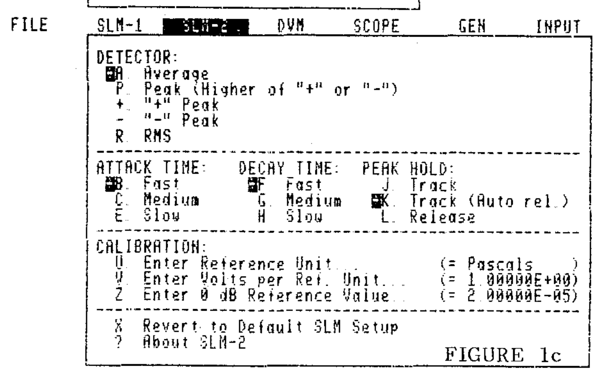


FIGURE 1c

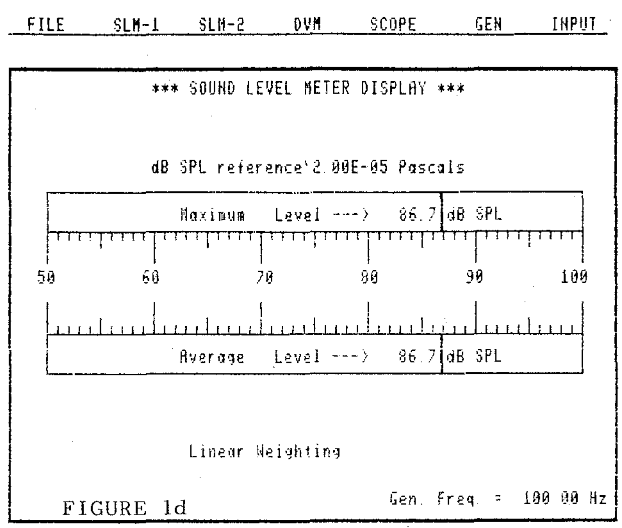


FIGURE 1d

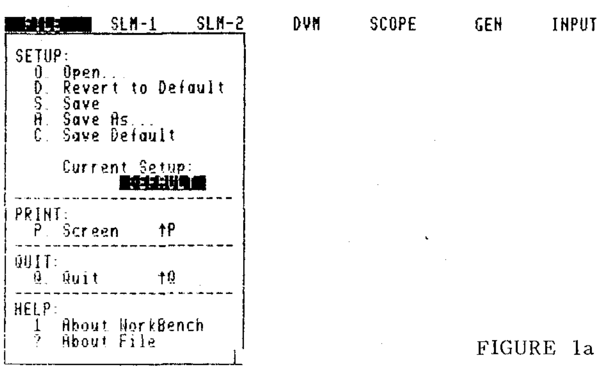


FIGURE 1a

```
FILE   SLM-1   SLM-2   DVM   SCOPE   GEN   INPUT
DISPLAY:
  0. Display Digital Volt Meter
--------------------------------------------------
FUNCTION:
  D  DC
  A  AC Average (RMS Indicate for Sine)
  P  AC Peak
  I  Impedance (Mag-Phase & Real-Imag)
--------------------------------------------------
ATTACK TIME:   DECAY TIME:   PEAK HOLD:
  B  Fast        F  Fast       J  Track
  C  Medium      G  Medium     K  Track (Auto Rel)
  E  Slow        H  Slow       L  Release
--------------------------------------------------
SCALE (FS):                  EXTERNAL ATTENUATOR:
  1.   2 Volts                 M  Off (X1)
  2.   1                       N  On  (X0.01)
  3. 500 mVolts
  4  200                     ZERO CENTER for DC:
  5  100                       S  Off
  6   50                       T  On
  7   20
  8   10
  9    5
  0    2
--------------------------------------------------
  X  Revert to Default DVM Setup
  ?  About DVM
```

FIGURE 1e

FILE SLM-1 SLM-2 DVM SCOPE GEN INPUT

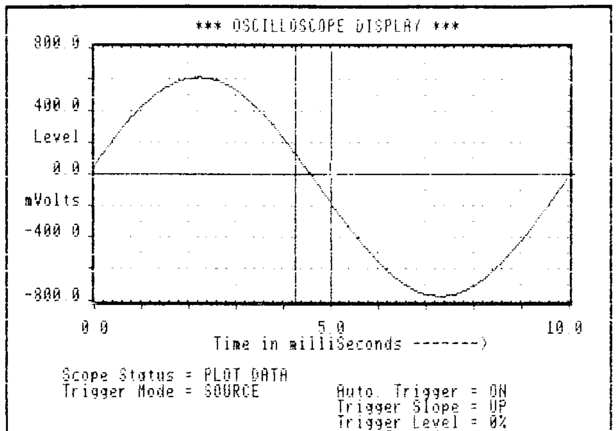


FIGURE 1h

FILE SLM-1 SLM-2 DVM SCOPE GEN INPUT

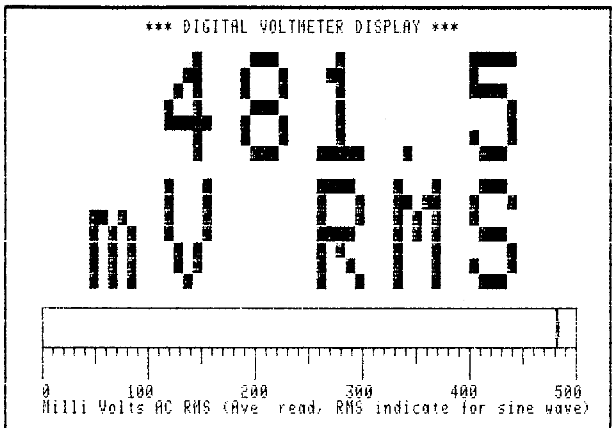


FIGURE 1f

```
FILE   SLM-1   SLM-2   DVM   SCOPE   GEN   INPUT
DISPLAY:
  0. Display Function Generator
--------------------------------------------------
FUNCTION:               MODE:
  S. Sine                 H  Fixed Frequency Mode
  T. Triangle             I  Continuous Sweep Mode
  W  White Noise          J  Triggered Sweep Mode:
  N  Pink Noise           K     Start Sweep
                          L     Stop (Abort) Sweep
  Q  Output On            P     Pause Sweep
  R  Output Off           M     Continue Sweep
--------------------------------------------------
OUTPUT VOLTAGE:         SWEEP TIME:
  1.   2  V RMS           A  50 Secs
  2.   1                  B  20
  3. 500 mV RMS           C  10
  4  200                  D   5
  5  100                  E   2
  6   50                  F   1
  7   20                  G   0.5
  8   10
--------------------------------------------------
FREQUENCY:
  U  Enter Fixed Freq  (Hz).  (=   100.00   )
  V  Enter Start Freq  (Hz).  (=   100.00   )
  Z  Enter Stop  Freq  (Hz)   (= 10000.00   )
--------------------------------------------------
  X  Revert to Default GEN Setup
  ?  About GEN
```

FIGURE 1i

```
FILE   SLM-1   SLM-2   DVM   SCOPE   GEN   INPUT
DISPLAY:
  0. Display Oscilloscope
--------------------------------------------------
HORIZONTAL SWEEP TIME:        VERTICAL SENSITIVITY:
  1.   2  Secs/Div              A    1  Volt/Div
  2.   1                        B  500  mVolt/Div.
  3. 500  mSecs/Div.            C  200
  4  200                        D  100
  5. 100                        E   50
  6.  50                        F   20
  7.  20                        G   10
  8.  10                        H    5
  9.   5                        I    2
  0.   2                        J    1
  Y.   1                      EXTERNAL ATTENUATOR:
  Z. 540  uSecs/Div             M  Off (X1)
                                N  On  (X0.01)
--------------------------------------------------
SWEEP TRIGGER:                           LEVEL:
 MODE:   !      AUTO:     SLOPE:           >  +75%
  9  Free Run     L  On     Q  Up          1  +50%
  S  Source       F  Off    T  Down        U  +25%
  K  Keyboard                              V    0%
                                           W  -25%
                                           [  -50%
                                           (  -75%
--------------------------------------------------
  X  Revert to Default SCOPE Setup
  ?  About SCOPE
```

FIGURE 1g

FILE SLM-1 SLM-2 DVM SCOPE GEN INPUT

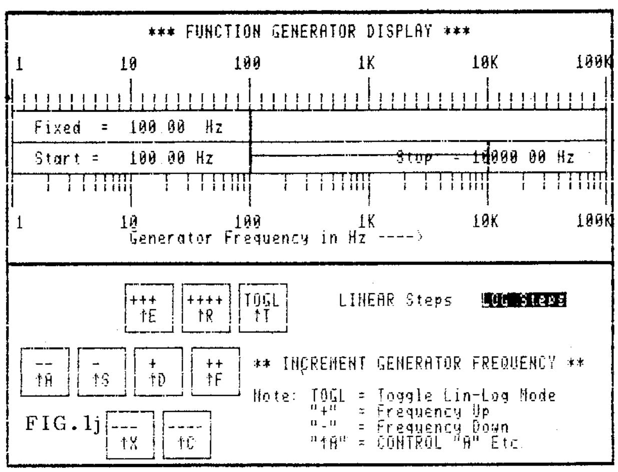


FIG. 1j

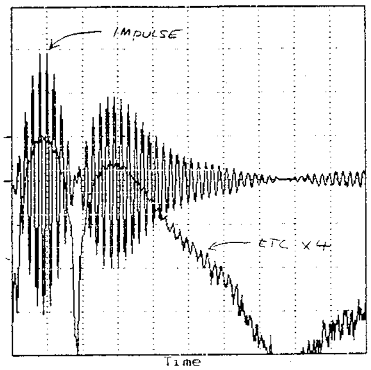
```
FILE   SLM-1   SLM-2   DVM   SCOPE   GEN   INPUT
CONFIGURATION:
  B  Both inputs on (Balanced, normal)
  P  "+" on only (Unbalanced, normal)
  N  "-" on only (Unbalanced, inverted)
  0  Off

  D  Double Integration (Accel. to Disp.)
  S  Single Integration (Accel. to Vel. )
  Z  No Integration
------------------------------------------
  X  Revert to Default INPUT Setup
  ?  About Input
```

FIGURE 1k

WHY NOT IMPULSE MEASUREMENTS?

The question was asked during the Concert Hall Workshop at the October 1985 AES in New York City, "Why not use impulse measurements rather than the more complex ETC measurement for examining time domain behavior?" Heyser answered most succinctly. A bell like tone doesn't go like this (he waved his hand up and down), but like this (he moved his hand in a smooth decaying curve). The illustration shown here has overlaid the impulse response and the energy time curve response. The ETC is the result of the impulse being squared; the doublet being squared summed and then the square root taken for each frequency. The ETC shows what's truly going on.



VERTICAL:

Linear amplitude for impulse and 6dB for ETC.

HORIZONTAL:

0 microseconds or 0 FT to 1575 microseconds or 1.77974 FT

RIGHT HEMISPHERE - LEFT HEMISPHERE

R. W. Sperry has described the right cerebral hemisphere (for right handed persons) as "...seem to be holistic and unitary rather than analytic and fragmentary...and to involve concrete perceptual insight rather than abstract, symbolic, sequential reasoning."

H. Gardner has written:

> "Here the patient (with right hemisphere damage) exemplifies the behavior...associated with the brilliant young mathematician or computer scientist. This highly rational individual is ever alert to an inconsistency in what is being said, always seeking to formulate ideas in the most airtight way; but in neither case does he display any humor about his own situation.... One feels rather that the answers are being typed out at high speed on computer print out paper."

For further study, see Philip J. Davis and Reuben Hersh's "The Mathematical Experience," Houghton Mifflin Company, Publisher; section six, the chapter entitled Nonanalytic Aspects of Mathematics, pages 314-316.

A SET OF BASIC ACOUSTIC EQUATIONS

In order to visualize the basic acoustic parameters we utilize in sound system design, we start with an imaginary point source that radiates omnidirectionally. If it has a peak amplitude "A" of 0.00116" (0.029464MM) at a frequency "f" = 1000 Hz from a cone with a diameter "D" = 10" (254MM) and we measure at 0.925' (0.282M), from this data, we can calculate the acoustic power "Wa"

$$Wa = \left(\frac{\text{Apeak } f^2 D^2}{116{,}000}\right)^2$$

$$Wa = \left(\frac{0.00116\ (1000)^2\ (10)^2}{116{,}000^*}\right)^2 = 1 \text{ watt}$$

*S.I. = 1.9 for M

This is the total acoustic power without regard to distance or directionality.

Having the acoustic power Wa allows further calculation of the sound pressure P_{RMS}.

$$P_{RMS} = \sqrt{\frac{Wa\ Pc}{4\pi r^2}}$$

From our data given, we find:

$$P_{RMS} \sqrt{\frac{1w\ (406)}{4\pi(0.282M)}} = 20.15 \text{ pascal}$$

Acoustic power and sound pressure can be converted into sound power level L_w and sound pressure level L_p. By defining an area we can also convert sound power into sound intensity and sound intensity level.

$$L_w = 10 \text{ LOG} \left(\frac{Wa}{10^{-12}w}\right)$$

$$L_w = 10 \text{ LOG} \left(\frac{1w}{10^{-12}w}\right) = 120 \text{ dB}$$

If we use our given distance from the source of 0.282M and assume omnidirectionality then we are able to calculate the acoustic intensity.

$$4\pi\ (0.282M)^2 = 1.0M^2$$

Therefore, our 1.0 Wa can be said to be: $1.0w/_M2$ at the surface of our sphere defined by our radius of 0.282M.

The acoustic intensity level is then: $$10 \text{ LOG} \left(\frac{Wa/M^2}{10^{-12}w/M^2}\right)$$

$$10 \text{ LOG} \left(\frac{1w/M^2}{10^{-12}w/M^2}\right) = 120 \text{ dB}$$

The reference for sound pressure level calculations was chosen to allow a similar number to be obtained under the same circumstances as above.

$$20 \text{ LOG} \left(\frac{P_{RMS}}{0.00002pa}\right)$$

$$20 \text{ LOG} \left(\frac{20.15pa}{0.00002pa}\right) = 120dB$$

In real life problems, we encounter two major variables that affect our every calculation, namely, the directivity factor "Q" and the distance from the source "Dx".

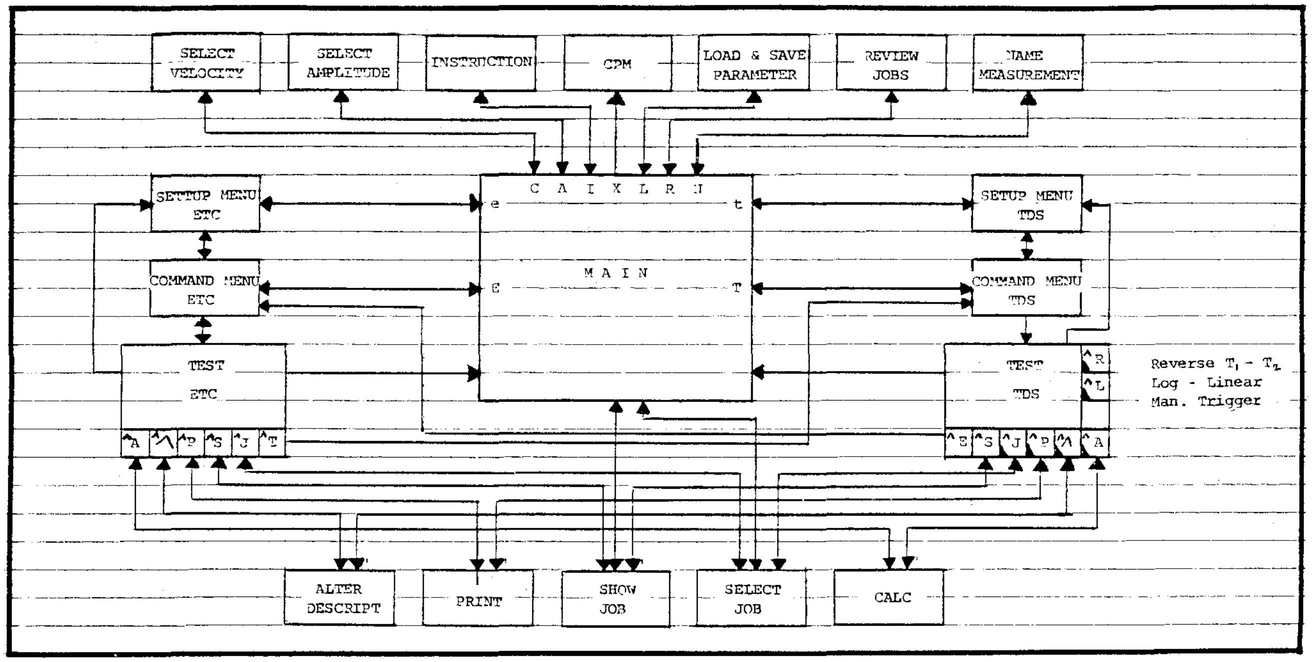
$$L_p = Lw + 10LOG \left(\frac{Q}{4\pi(Dx)^2}\right) \qquad \underline{\text{OUTDOORS}}$$

$$L_p = Lw + 10 \text{ LOG} \left(\frac{Q}{4\pi(Dx)^2} + \frac{4}{S\bar{a}}\right) + 0.2 \qquad \underline{\text{INDOORS}}$$

Other variations on this theme are for semi-reverberant spaces, accounting for the frequency dependency of the equations, and accompanying influences, such as, the delay behavior of signals from "real life" arrays. These derivations do allow a conceptual view to be formed for later more detailed expansion.

TEF SOFTWARE ROAD MAP

Jeff Long of HIS Sound in Fresno recently sent us this really excellent layout of the TEF analyzer's software commands for measurements. Road maps of this type are invaluable when you are in the midst of a measurement and can't go back to the internal instructions in the machine without losing the measurement.



CLEARWATER - OCTOBER 30-31, 1985

SOUND & VIBRATION EDITORIALS

Two valuable, very much to the point, editorials have appeared recently in **Sound and Vibration** magazine. The first by Donald R. Houser was in the Sept. 1985 issue and was entitled "Signal Processing With The Personal Computer--A Warning."

Mr. Houser remarks, "Although the processing is typically quite slow (with software FFT algorithms) relative to dedicated spectrum analyzers, this approach to signal processing would **seem** to be a real bargain."

"But, BEWARE!! Successful spectrum analysis requires more than a good FFT algorithm....In order to obtain **valid** spectra, one must first use an anti-aliasing filter... So you say, let's go out and buy an anti-aliasing filter...it will have a price tag comparable to your P.C."

"The 120 dB/octave low-pass filter in most spectrum analyzers allow valid (free of aliasing over a 40 dB dynamic range) spectra over 400 of the 512 lines which are computed with a 1024 point FFT... A 48 dB/octave filter only allows 280 valid lines."

There is much more detail on all these matters in this editorial and we highly recommend that anyone tempted by FFT software for a P.C. arm themselves with this information.

The second editorial is by George Fox Lang in the Oct. 1985 issue of **Sound and Vibration,** entitled "Mighty Is The Power Of The Sine."

"Impulse testing maximally taxes the tester, the tested, and everything in between.... The resulting high crest-factor data is a violent test of the dynamic range of the entire measurement system from sensor to spectrum computation."

"FFT analyzers...are singularly ineffective for the time efficient implementation of a swept-sine study and few users employ them in such a fashion more than once."

While this author is addressing the needs of modal studies in the vibration field, the truths he enunciates are valid for all measurement systems. Two places in particular where TEF testing is completely superior to impulse or burst testing is in the measurement of loudspeakers and the analysis of acoustic environments. Both of the above benefit in remarkable ways from a testing system with the maximum possible signal-to-noise ratio.

We are particularly pleased to see magazines, such as **Sound and Vibration,** providing a forum for responsible well-informed writers to uncover the facts needed if we are to intelligently acquire and utilize the new technologies.

Sound and Vibration is free to qualified people. Write to: 27101 E. Oviatt Road, Bay Village, OH 44140.

TECH TOPICS VS NEWSLETTERS

We are changing the size of future issues of the Newsletter, making it fewer pages. We will increase the number of Tech Topics so that the number of pages of each quarterly mailing is approximately the same as in the past.

We are making this change so that information on specific subjects is readily available for future reference.

CLASSIFIED

FOR SALE: Crown RTA-2 Audio Spectrum Analyzer, like new, less than 50 hours operation, purchased 1981 - $2,000. Asking . . $500.00 or best offer.

CONTACT: Jim Stinson,
c/o VJ Engineering
P O Box 8463
Newport Beach, CA
(714) 548-1622 (H)
(714) 548-7334 (B)

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Gaussmeter, Bell 600 with small probe $300.00

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(916) 338-2097

FOR SALE: HP 41C with Quad Memory Module, Printer, Wand, Card Reader, PPC ROM, Home Management ROM, Extra magnetic cards, new box of black printing paper, Audio 41-C programs, HP User Club programs and others. $650.00

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3020 Balboa St.
San Francisco, CA 94121
(415) 386-7154

POSITION WANTED:

Mohamed A. Youssif has a B.S. in Telecommunications and Electronics Engineering from Helwan University, Cairo, Egypt, May 1983. His design project: Alarm System Controlled by Microprocessor, with a grade of very good.

Mr. Youssif is eager for work in communications and will relocate. He can be located at: 4342 Carmelo Drive, Apt. 102, Annendale, VA 22003. Phone: (703) 256-7985.

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