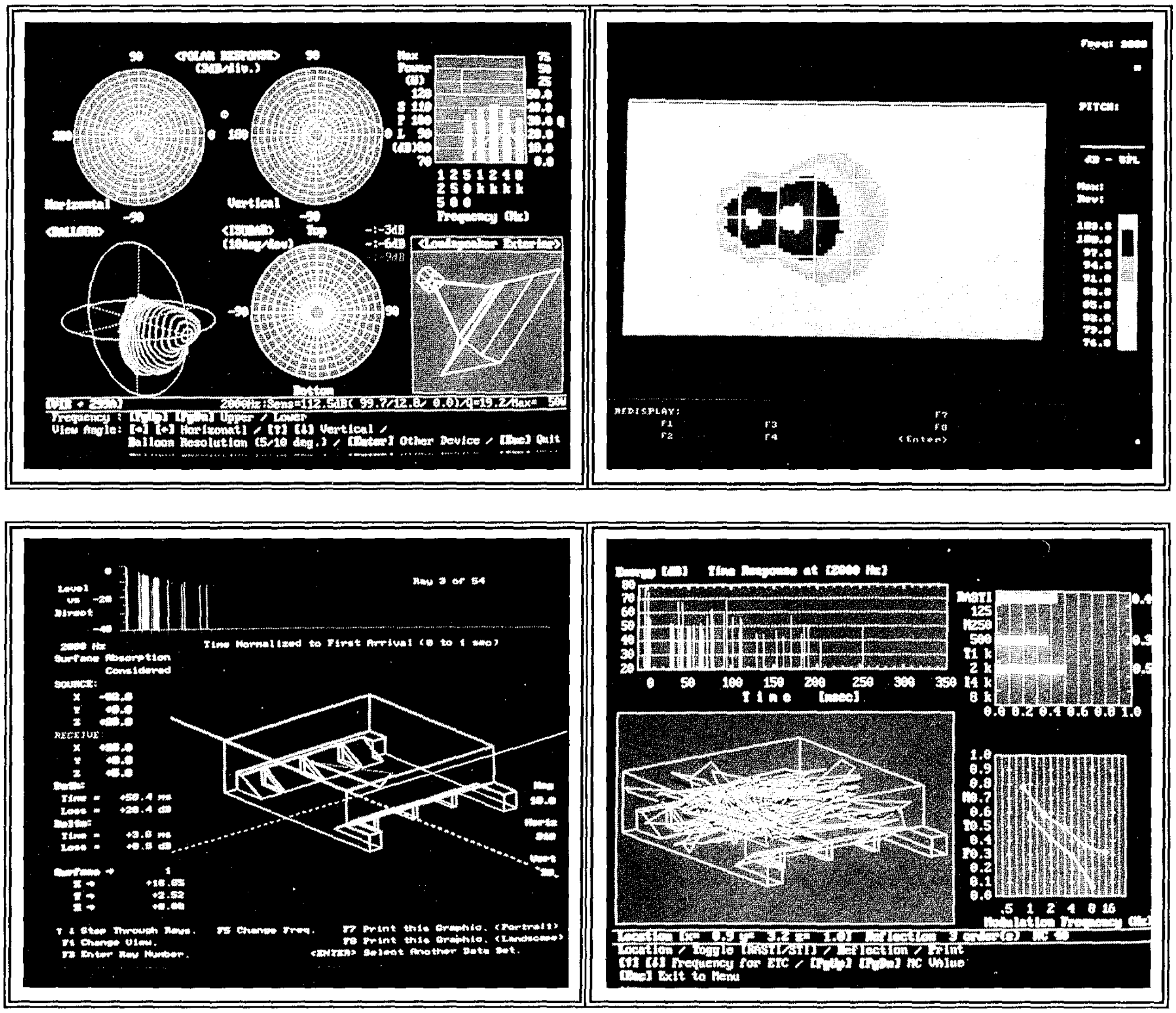
SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

newsletter

Volume 19, Number 1
FALL, 1991

Altec Lansing



Modern analysis allows visualization and auralization of computer assisted designs.

SYNERGETIC
SYN AUD CON
AUDIO CONCEPTS

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

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.

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TABLE OF CONTENTS

When Do I Renew?

You can check to see when your subscription will expire by checking the mailing label on the envelope in which your newsletter was mailed. In the upper righthand corner, beside the name, a date will appear (i.e. 4-90). This means that you will receive this issue and it will be the last issue sent unless you renew. Renewal notices will be sent at this time. You must renew before the next quarter's newsletter is mailed or your subscription will become inactive.

POSTMASTER: Send address changes to Synergetic Audio Concepts, R.R. 1, Box 267, Norman, IN 47264.

Medora and the Medora High School Gymnasium could have been used for the movie Hoosiers. But, the gym has serious problems: 3 seconds at 2,000 Hz and an NC curve around 70 dB. We had fun showing the gym to visiting sound men, and the common statement was, "Give them my competitor's card. It will keep him off the streets for months." That all changed when Jerry Spriggs of Altec had a vision when we took him to see this "troubled gym". He saw it as a golden opportunity to have a workspace capable of testing the modern tools of analysis and design. To verify the accuracy of speech intelligibility measurements, you must first have an environment that seriously degrades the desired intelligibility.

He proposed to Altec President, Dave Merrey, that Altec donate the equipment to install two different distributed systems over the bleachers and a VIR with woofer to cover the gym floor plus all the needed electronics.

Remarkable things began to happen when we told others of Altec's generous donation: Pat Brown of Pro Sound Audio about an hour away in Clarksville offered his time, his truck, and tools with a magnificent Genie scaffolding. Fred Fredericks from San Diego was visiting us and he donated many days with Pat installing the sound system so that each speaker in the distributed system is a "home run" to the rack room with complete access individually to every component for measurement purposes. Each valid design approach can be measured, listened to, and adjusted.

Soundolier supplied the rack equipment as a donation to the school; West Penn Wire donated the wire and cable; and Shure Brothers, EV and Crown donated microphones for specific needs. Earlier Ron Steinberg of RentCom in Chicago had donated a pair of multicells and the original system consisted of a Soundsphere. This meant that there were now five individual sound systems for members of a workshop to study.

The cover depicts the additional contributions of Altec. In a very short time Gary Jones, Jack Vig, and Akira Mochimaru had made AcoustaCadd sources directional and the walls variable absorption and, therefore, were able to produce good quality auralization of the gym from the drawings. The balloon shown is for Altec's new VIR horns and the predicted coverage exactly matched that achieved on the floor of the gym.

On August 5-7, 1991, 30 dedicated professionals, all of whom were deeply into modern analysis and design, came to this Mecca to worship and learn.

We will be writing a great deal more about the August 1991 Workshop in our next Newsletter. We consider it one of the most important workshops we have participated in. We wanted to let you know now who the people are that made it all possible. The tiny rural community of Medora, fiercely independent of the consolidated school system, were the recipients of the remarkable and generous outpouring of Syn-Aud-Con sponsor and friends. Our heartfelt and sincere thanks and gratitude to each and everyone of them.

1991 Professional Audio Directory

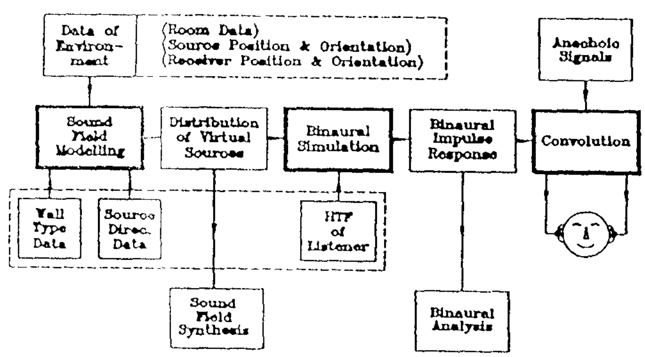
Hannah Klapholz (Jesse Klapholz's wife) has published a 136 page "1991 Professional Audio Directory."

This is a very comprehensive directory and is indexed in such a manner as to speed up locating manufacturers, products, and acoustical and design consultants. If you are not included and should be, please be sure to be in touch with them.

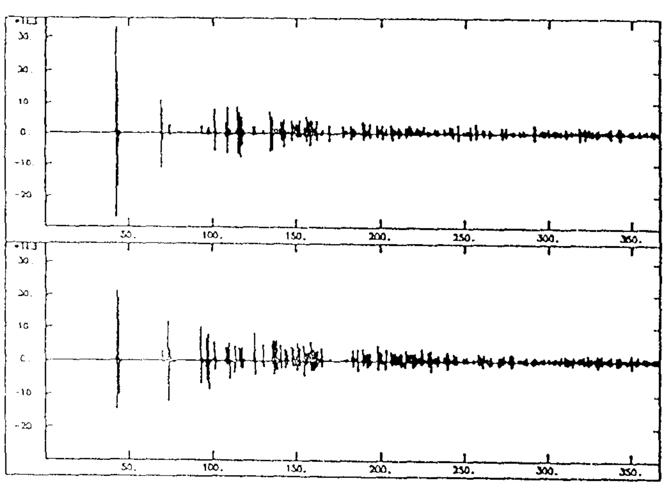
Formatted as an Audio Yellow Pages, each section has a useful company vs. product cross reference chart; followed by the company listings which include both sales and engineering contacts for each company.

The directory is the best we have seen, and it is FREE to anyone in the professional audio industry. Anyone interested in becoming a subscriber can simply send a postcard, with a signature and date requesting the directory to: Professional Audio Journals, Inc., PO Box 31718, Philadelphia, PA 19147-7718. PH: 215-465-1975, FAX: 215-336-7743.

Professional Audio Journal welcomes comments and suggestions. When you get your free copy, let them know what you think of it; we think it's great.



Schematic of the Bochum Binaural-Room-Simulation System



Binaural impulse response (left ear is upper picture).

Binaural Auralization

H. Lehnert and J. Blauert have written a superb paper entitled, "Virtual Auditory Environment". In this paper they state:

> "The simulation process can be split up into two computational stages, namely, sound-field modeling and auralization."

In this paper they outline in good detail a complete binaural auralization system that includes surface data, source directivity data, the head transfer function of the hypothetical listener in the unbuilt room -- all combined into a sound field synthesis in the form of a binaural impulse response able to be convolved mathematically with anechoic music or speech sources for playback over headphones or loudspeakers.

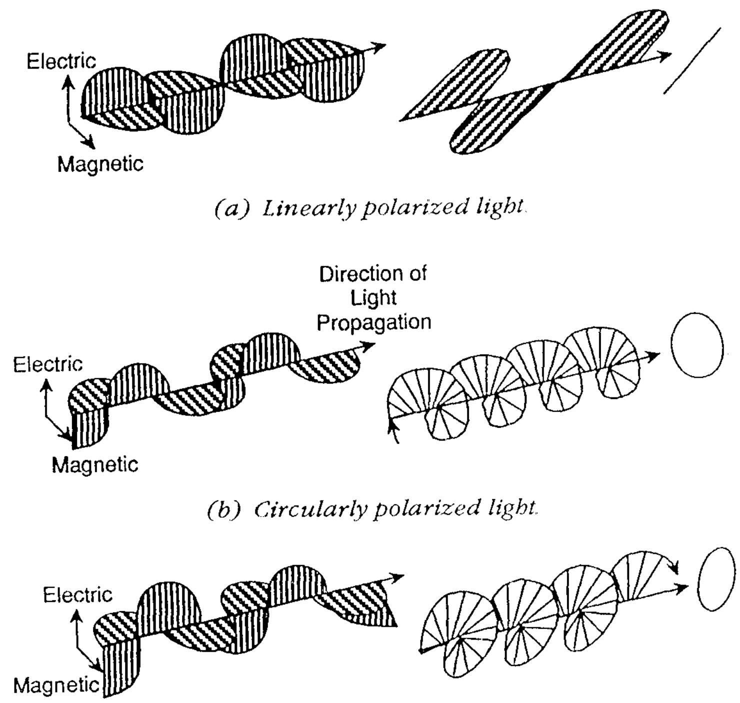
Hopefully the paper given by Dr. Blauert in New York at AES will be similar to the paper we have and be available in preprint form.

Fiber Optics Polarity

The electromagnetic field is just that: an electric field and a magnetic field related to each other by a Hilbert transform.

Light is an electromagnetic phenomenon. Light can be linearly polarized, circularly polarized or by means of phase Ellipitically polarized.

In acoustics we have a 180° polarization and a 90° polarization (the Hilbert transform of the impulse response or the magnitude response in each of the two major domains - time and frequency).



(a) Linearly polarized light.

(b) Circularly polarized light.

(c) Elliptically polarized light.

Types of polarized light.

From Advanced Digital Audio, editor Ken Phohlmann

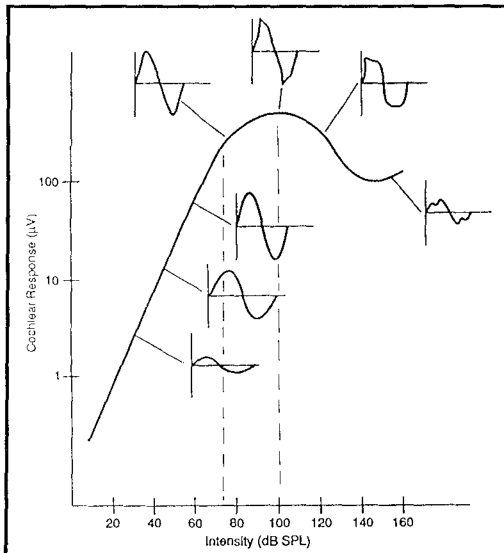
How Loud is too Loud?

We found the following, very interesting plot in a new book offered by Sams, entitled **Advanced Digital Audio** by Ken Pohlmann, editor.

In the chapter, "Human Auditory Capabilities" by Daniel C. Mikat there is the statement that for home systems a dynamic range of 101 dB is required for "audible perfection" and 125 dB for professional system perfection. If this is true it would appear that professionals prefer more distortion than do consumers.

I have drawn in two dashed lines, one for an LP of 70 dB and the second for an LP of 100 dB. At 70 dB the first visible waveform distortion is apparent. At 100 dB the distortion is gross at 120 dB the human listener is square waving.

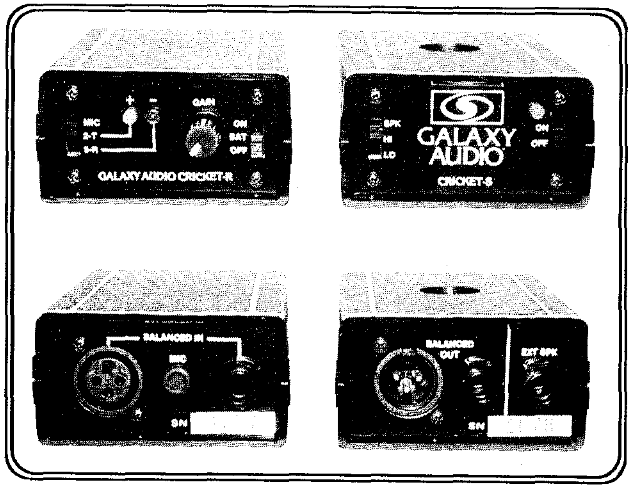
Forget hearing safety, forget audience annoyance (non-rock audience, that is), forget neighbor complaints; just recognize that these levels present a completely distorted aural view of reality.



Courtesy Daniel C Mikat

It seems to me that the only critical judgment that one can make on an A-B test at 120-130 dB is whether the systems rattle or not. It certainly isn't going to be possible to make any quality judgments.

"Cricket" Polarity Checker Available



Galaxy Audio of Wichita, Kansas now offers a new version of polarity checker. Their new "Cricket" consists of a "send" unit and a "receive" unit. Using XLR and 1/4 inch connectors for balanced circuits and providing either pin 2 or pin 3 high makes this a practical useful instrument. One feature usually not available is a gain control. We are told that the suggested user price is under $300. Housed in high impact ABS cases and supplied with a padded carrying case this appears to be an outstanding bargain for an indispensable tool. Powered by a single 9 volt battery in each unit. Containing a built-in microphone in the receive unit and a built-in loudspeaker in the send unit allows acoustic polarity as well as electrical polarity to be determined.

We're using the Cricket in our classes now. It comes with a 5-page manual explaining all the uses for the Cricket.

For more information contact Galaxy Audio, 625 E. Pawnee Ave, Wichita, KS 67211, PH. 316-262-2852 or 800-369-7768; FAX 316-263-0642.

Sum Through Crossover

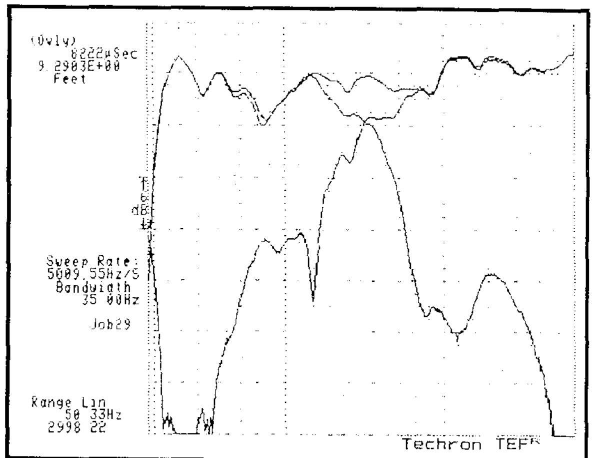
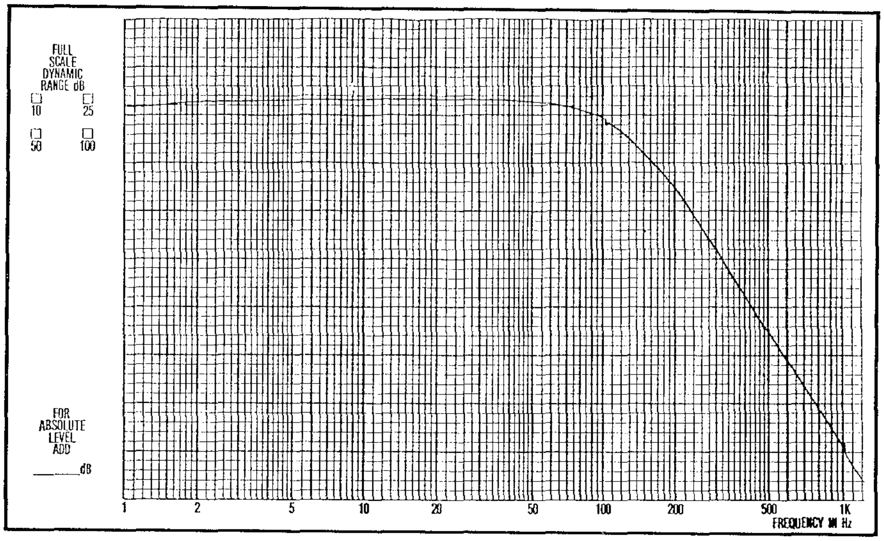


Fig. 1—How is this for a real world example? Gene

Gene Patronis is easily the King of Crossovers. He has demonstrated this repeatedly at our loudspeaker design workshops and in successful products for many manufacturers. We are currently engaged in converting our "in-the-ear" playback system from UREI monitors to Frazier "Cat" systems. (Not because we are dissatisfied with the UREI, but because it's time to return them from the loan.) The Frazier "Cat" units are fully signal aligned and are the result of the design skills of Jay Mitchell (a graduate of Georgia Tech and a former Patronis pupil). We asked Gene Patronis to design us a high quality low-pass network for 100 Hz for the signal going to the Intersonics woofer we use with this system. The following two illustrations show Gene's latest sum through network (for a new system he is designing). That's the acoustic response you're looking at and second illustration is the electrical response curve of our LPF.



Digital Filter Questions

Question: Can a digital filter change amplitude without changing phase?
Answer: Yes. Infinite Impulse Response filters (IIR) can

Question: Is this desirable?
Answer: Not if what you are trying to correct with the filter has a phase response that needs correction. Loudspeakers are the major class of devices we deal with that require that the phase response of the filter be the conjungate of the phase response of the loudspeakers in order to properly correct amplitude variations without introducing extraneous unwanted effects.

Question: What authorities say this?
Answer: Starting with Dr. Stockham back in the 70s; Dr. Patronis, and recently Dr. Cabot - all agree that a minimum phase response filter is required to correctly equalize a minimum phase response transducer.

Oppenheim and Schafer in their excellent text, Digital Signal Processing have the following remarks about IIR filters vs FIR (Finite Impulse Response) filters (From pages 268-269)

"The examples of the previous section make it clear that IIR filters generally achieve excellent amplitude response at the expense of non-linear phase (very non-linear at band edge). IIR filters are less costly and require less computational ability on the part of the designer.

"There is an Optimality Theorem for FIR filters that is meaningful in a wide range of practical situations. This means that the design of FIR filters is under more control than IIR filters."

Question: Does this mean that IIR filters are the cheap and dirty way to build a digital filter?
Answer: Perhaps - unless someone has found a better way than the texts describe.

September Farm Seminar 1991

Syn-Aud-Con 1992
Seminar & Workshop Schedule

❖ 3—Day Seminars—$525
Farm—Norman, IN

May 21-23
June 18-20
July 16-18
August 20-22
September 17-19
October 15-17

❖ 3—Day Workshops—$650
Orange, CA
Concert Sound Reinforcement
January 14-16

❖ 3—Day Workshops—$600
Medora, IN
Auralization
May 14-16

❖ Concert Sound Reinforcement ❖
Today and Tomorrow
A Hands-on Look at Technology - Current and Developing

Orange County, CA
January 14-16, 1992

Fee: $650

Staff:

Will Parry, Chairman
Maryland Sound, Baltimore, MD

Albert Leccese
Audio Analysts, Colorado Springs

M L Procise
Showco, Dallas, TX

Mick Whelan
Electrotec, Canoga Park, CA

Roy Clair
Clair Brothers, Lititz, PA

David Scheirman
Facilities Coordinator
Concert Sound Consultants,
Julien, CA

There is magic in watching the greats of our touring sound reinforcement industry participating in a truly synergetic sharing. These men are very competitive, yet they come together as friends and work together as a remarkably synchronous team.

This year the Workshop will be co-sponsored by ProSound News and Syn-Aud-Con. We are very grateful for the support from the press for this important Workshop.

The Workshop will be held at Chapman College in Orange, CA. Attendees will stay in Anaheim and be bussed to and from the Workshop.

All meals, including continental breakfast are included.

JBL 1991 AUDIO ENGINEERING CONFERENCE

In September we were invited to participate in the JBL Audio Engineering Conference. The Conference was under the guidance of Gary Hardesty who was president and owner of Audio Digital, which JBL recently acquired.

The JBL Conference is held for JBL contractors, consultants and end-users every other year and has personnel from all over the globe in attendance. This meeting was well attended and included a large number of Syn-Aud-Con grads both in attendance and on the staff at JBL.

The Conference can't fail to be a success when it brings together 300 outstanding individuals from our audio industry and allows them to "stir the pot" for 2-1/2 days! And it was a success.

JBL had Sam Berkow on hand to explain and demonstrate auralization as well as to consult with him on including auralization in their new computer program. The new CADP II works under Windows, which means that it will most likely be easy to use.

Doug Button presented a very informative discussion of which mechanical properties of a loudspeaker generate which acoustical distortion components.

A gentleman by the name of Harry Donovan, a master rigger, presented a remarkable lecture on the dangers, liability and prevalence of poor rigging practices. If I'm ever to rely on the rigging, a doubtful possibility considering my acrophobia, it would be Harry Donovan or no Don.

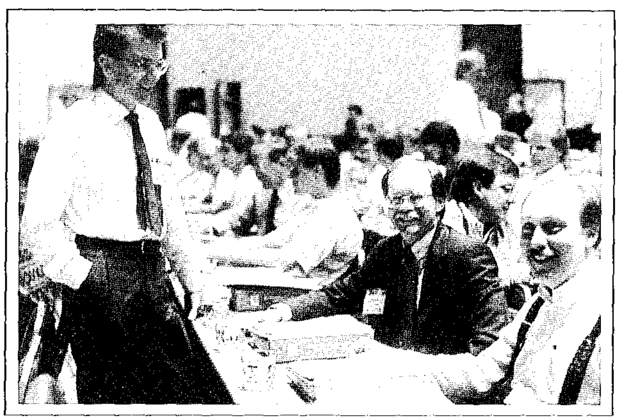
Bob Adams of Hoover & Keith gave a detailed description of an innovative program he uses to train the operators of sound systems. His methods appealed to us as universally applicable and useful.

Craig Janssen has joined SIA - Sam Berkow's consulting firm in Dallas (by the way, SIA means Sam I AM) came across in his part in the JBL program as the talent that he is: dynamic, aggressive, questioning, probing— determined to design the finest sound system that he can with the dollars available to him.

The JBL staff literally exudes genuine interest in their contractors and invited consultants.

New products included a packaged system (small format) for use in forming large array clusters. This has been the project of Bill Gelow, now with JBL after 10 years at Renkus-Heinz.

A truly dynamic and humorous guest lecturer, Dr. Warren Bennis, was the before-dinner speaker for the first evening. His discussion of the difference between leaders and managers acquired and kept our attention throughout his talk. We were so impressed with his talk that we bought one of his many books, On Becoming a Leader. I especially wanted to read in more detail his discussion of the difference between managers and leaders. We will write more on this later, but one of the differences he mentions is: A good leader does the right thing; a manager does things right.



His story about being in a workshop with Henry Kissinger when Kissinger was asked "How different do you think things would have been if Kruschev had been assassinated instead of Kennedy?" And Dr. Kissinger replied, "Well, he is sure of one thing, Mrs. Kruschev wouldn't have married Onasis."

Finally, yours truly gave a talk on TEF, in-the-ear microphones, and auralization and how they relate to each other.

One lecturer made a point of stressing how necessary 600Ω was to the dBM. Carolyn provided emergency restraint to Don, but he was pleased at how many grads came up to him later and mentioned their own surprise that such errors were still prevalent.

Meetings like this one are invaluable aids in product training and in getting to know your colleagues in the business.

The JBL staff literally exudes genuine interest in their contractors and invited consultants. We were pleased to be a part of such a successful meeting in the midst of our friends and Syn-Aud-Con grads.

The First Baby Steps with the Heyser Spiral

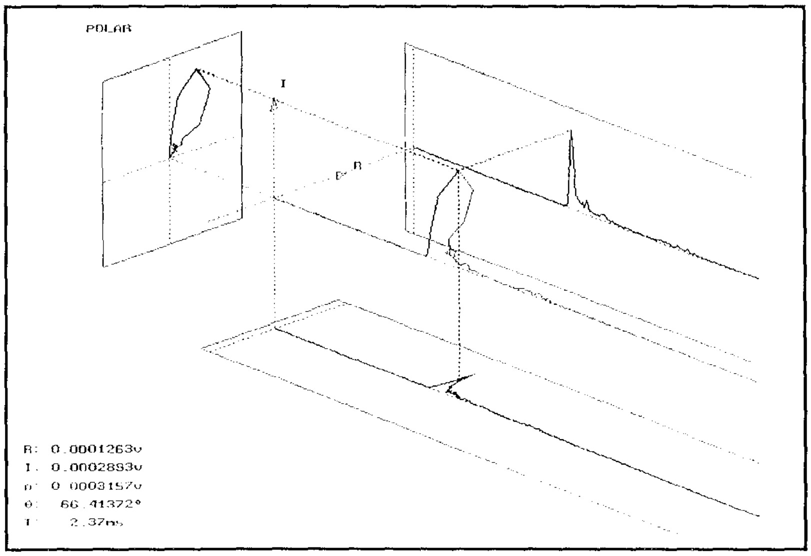


Figure 1

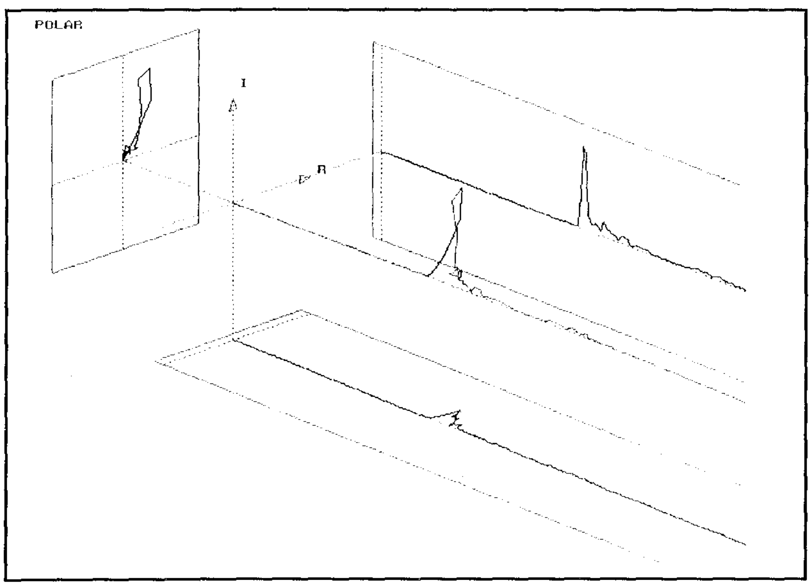


Figure 2

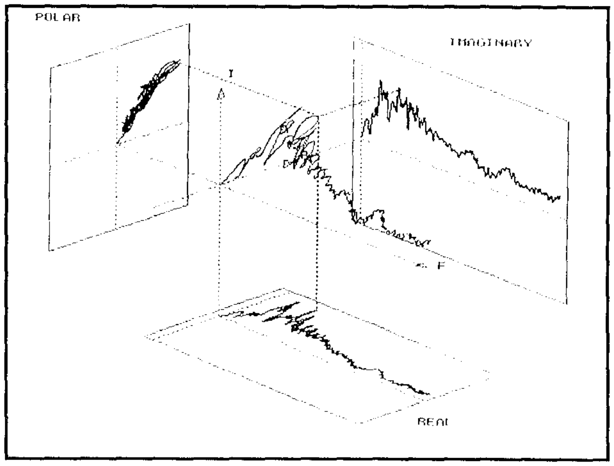


Figure 3

Sometimes when a new concept is presented someone will ask "What use is it?" A wit once answered "What use is a new baby?"

When we first demonstrated the new Heyser Spiral program someone asked, "What good is it?" Well, the baby has taken its first steps.

Figure #1 is taken with a SASS and the real part (bottom) is the left ear and the imaginary part (top) is the right ear. The Nyquist polar and the Spiral are the partitioning between the two ears in the time domain.

Figure #2 is also in the time domain but now the real part is the sum of the two ears and the imaginary part is the difference. Remember the people who used to challenge Dick about the phase of time. It sure shows up in the sum. It immediately becomes obvious that the difference is a significant parameter in the time domain (and probably a directional clue as well).

Continued on the next page. . .

Figure #3 is the same situation but now viewed in the frequency domain. The real is the left ear, the imaginary the right ear.

Figure #4 is the frequency domain view of the sum (real) and the difference (imag.) For this signal and this receiver position the part of our brain that listens to frequency probably uses the sum.

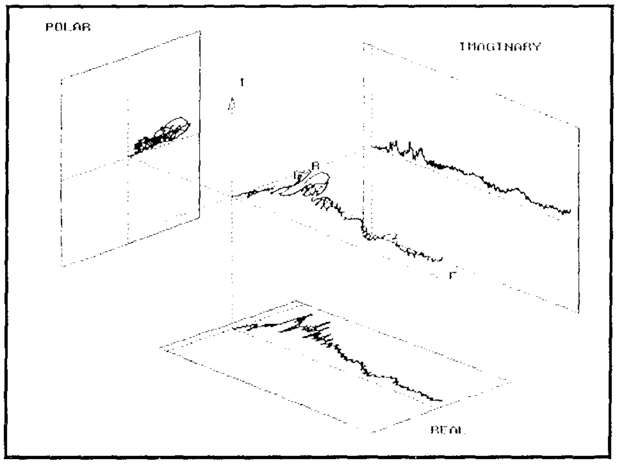


Figure 4

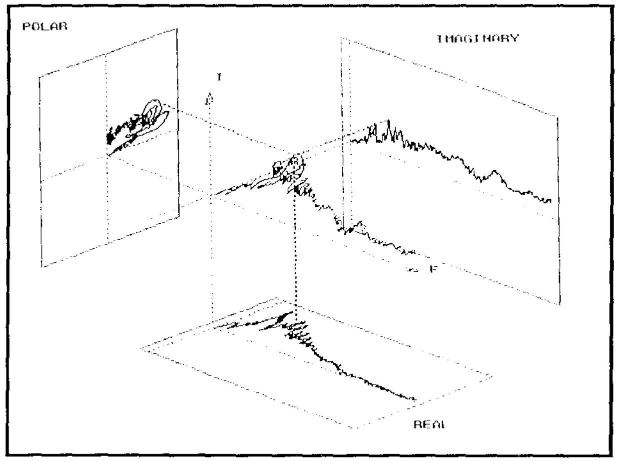


Figure 5

Figure #5 is exactly the same case in the frequency domain but with the time window made larger so that many early room reflections are included. Note the increased randomness the reflections introduced and the Nyquist polars familiar look from the days when x-y oscilloscopes were used to look at the sum and differences in stereo signals.

We can tell you with certainty that you will be hearing a lot more from this baby as it matures.

Auralization Report

The Fall AES convention had a special session on Auralization. Sam Berkow's paper was "lost" and therefore his name did not appear on the official program. He was allowed to present his paper last.

Sam's first public demonstrations of auralization (music and speech mathematically convolved with room data) to the audio industry, and to our knoweldge these are the first on a PC, were at the Fall 1990 AES Convention, the Syn-Aud-Con Intelligibility II workshop, and by Sam and ourselves at the 1991 NSCA convention and at our Advanced Acoustic Measurements workshop in August, 1991. Sam would like me to point out that the first auralization that he heard was by John Walsh of Artec, Vancouver, BC, on a mainframe computer.

There are a lot of people and companies playing catchup and all due credit to them, but Sam led the way on this side of the pond.

Sam felt that the French had done advanced research on this subject as manifested in a paper by Dr. P. Martin. Mandel Clymer from Denmark is another active developer of auralization techniques.

Binuaral was the key word for the majority of these auralization papers. Dr. Ahnert's presentation of a binuaral auralization package for use with Renkus Heinz' EASE program included the head transfer functions for use with the "probe". As of this writing only EASE has demonstrated their own auralization package. While others are in the works using Hypersignal, this package from Dr. Ahnert represents a remarkable feat in the development of an entirely new program.

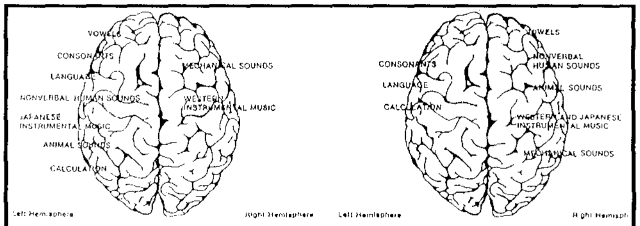
One manufacturer has suggested that the users of their design program send the design data to them and they will return a recording of the auralization. We suppose if you can take their design program on blind faith, why not a blind auralization.

We will continue to demonstrate to every class here at the farm the usefulness and excitement of this new tool.

Further Thoughts on Ear/Brain Processing

The illustrations shown in Figure 1 are familiar to the readers of **Sound System Engineering**, second edition. They appear on Pages 8 and 9. We have had the occasion in recent months to rethink what we think we know about these diagrams.

For instance, we have always been puzzled why the experimental evidence at hand suggests that we integrate over a longer period for noise or loudness than we do for speech intelligibility. We were then told by a researcher at Ariel (the manufacturer of the dual channel FFT board we are using both for measurements and auralization) that loudness is a right hemisphere function whereas speech intelligibility would be a left hemisphere function. Wham! Insight roused from its dormant state of thinking.



(A) Japanese pattern (B) Western pattern

Tadanobu Tsunoda, a specialist in hearing difficulties, believes that the two halves of the Japanese brain process sounds differently from Western brains. The key to those differences, he postulates, is that the Japanese deal with all vowels in the left hemisphere, while Westerners handle vowels in the right hemisphere

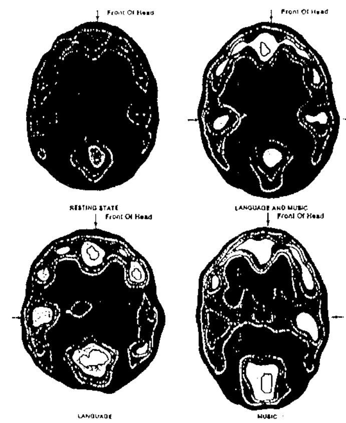


Figure 1

When you begin to look at the human brain as part of the processing, a great deal of unexpected and hitherto unsuspected data comes to light. While we can observe everyday the remarkable differences between humans, their sensory capabilities and their motivation are not directly observable. A recent measurement of brain responses reveals what has been termed an "Evoked potential." Fig. 2 shows one "trigger" mechanism that evokes the participants potential.

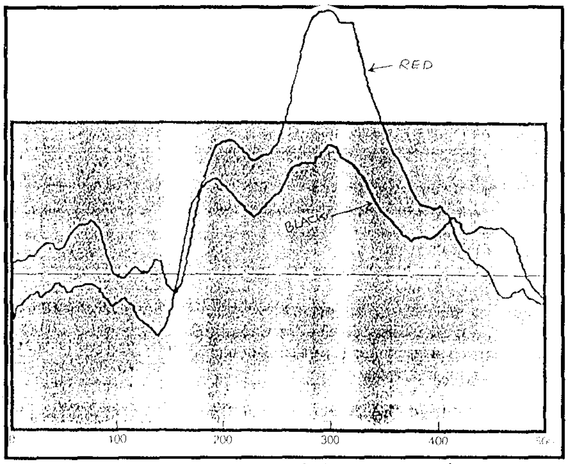


Figure 2. Evoked potentials of a subject who has just seen a number flash on a screen are shown in blue. In red, the subject's response to the same stimulus, after he is told that he can earn one dollar every time the number appears. Notice the positive component that pops up at 300 ms. This component, Dr. Begleiter noted, is a measure of how significant the stimulus seems to the subject.

During our Intelligibility II Workshop all the participants had extensive hearing tests and ear examinations. The correlation between lack of symmetry in the outer ear mechanism. (re: pinna and ear canal) and directional activity was significant, suggesting that there is a form of auto correlation between the two hemispheres.

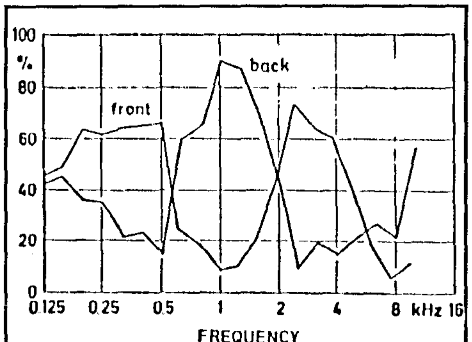


Figure 3

Diana Deutsch's work has shown that different frequencies receive definite preference from one ear rather than another when under transient excitation. This same frequency preference has been observed as Fig 3.

All of this would suggest that the sincere researcher would wish to be able to account for the following parameters in their "two eared" measurements.

1. The time domain and frequency domain signature of both ears as measured with head shadowing, pinna response, and ear canal resonance accounted for.
2. The difference between the signals arriving at the two ears.
3. The sum and the product of the signals arriving at the two ears.
4. The interaural cross correlation coefficient (IACC).

We soon will be working with the first three of the above. We will be reporting further on this research.

Re-Thinking Acoustic Measurements

When new tools arrive it is often time to re-think familiar measurements. Farrel Becker, at a recent Techron class, used the Heyser spiral to observe the interaction between resistance (real part) and reactance (imaginary part). Now, that's creative use of a new tool!

Reverberation measurements is another fruitful area for re-thinking what we do. Many simply want to imitate what we previously did with level recorders, in spite of the fact that TEF analysis has shown us that RT_{60} is, at best, a very coarse measurement and that early decay time EDT relates more meaningfully to the ear-brain of the critical listener.

Critical listening reveals that, even deep into the reverberant sound field in a statistically significant sound field, direct sound still plays a predominant role in detecting the direction to a source.

It would seem to us that there are two highly significant differences between live listeners and today's best instrumentation. The first, and perhaps most important, is that the listener is two channel.

Why We Need Two Channels

The human listener's remarkable ability to differentiate sounds in space is due in no small measure to having two reception channels, thus allowing differences and correlations to be detected.

A Heyser spiral plot of left ear vs right ear for correlations (real part) and differences (imaginary part) would have to be a measurement pregnant with new insights. The interaural cross correlation coefficient hints at the importance of such measurements.

Correct Microphony

The human listener is, at the eardrum, a directionally and frequency dependent receiver with distinct signal delay processing included; therefore, using omnidirectional microphones would seem to be remarkably unrelated to the listener's perception of the same signal. In-the-Ear, ITE, microphones would solve this fundamental oversight and when coupled with two channel analyzation equipment and a meaningful display, we believe there would exist a significant chance of increased correlation between what we hear and what we measure.❑

Fax for Help From Our Friends in Croatia

Just a year ago we were in Zagreb, invited by Ivan Stamac of Zagreb TV to participate in a Continuing Education program for ISOT.

Now we watch the destruction of the beautiful people of Croatia.

We received the following urgent call for help from Mr. Stamac. We sent the message to several of our grads that we know work with church organizations that may be able to help the children of Croatia.

If you personally send dried milk and soups, as we have, you will know that some children are going to have food because of you. Better yet if you can involve your church or an agency in making some contribution to their need.

"October 15, 1991

Dear Don and Carolyn:

In the State of Croatia, the mighty YU-Federal Army, hired by the Republic of Serbia have invaded so far more than 50% of Croatian territory. Following major cities are in the state of siege: Dubrovnik, Zadar, Sibenik, Pakrac, Osijek, Vinkovci and especially the border city Vukovar. In those cities, more than 1,000,000 people live between 15 and 50 days in the cellars of their mostly destructed homes. Above all, they need badly humanitarian help in: children's food, evaporated milk and vitaminized juices.

"I wonder if some of your local humanitarian organizations could send above said items to the following collecting point:

Klub "Donat"
c/o "Diadora"
Hotel InterContinental
Krxnjavoga 1
41000 ZAGREB, Croatia (YU)

"Phone numbers are +38 41 425 845 and +38 41- 425 945. FAX # is +38 41 424 952. Messages to myself are also possible to this fax number.)

"Do your best, dear friends, to help innocent children of this country. Any quantity of help will be welcome."

Innovative Use of Heyser Spiral

The most evident flaw in all the research into auditorium acoustics, that we see, is the use of conventional microphones. These environments are designed for human listeners, not omnidirectional microphones and omnidirectional sources. That the sources have significant frequency dependent directionality and magnitude variations is rarely considered. The listener likewise exhibits some remarkable and quite significant variations in discriminating either for or against certain directions, frequencies, and combinations of signals.

The human listener at some frequencies (below-1,000Hz) has a relatively uniform magnitude pressure response. At other frequencies they become both high Q receivers and frequency selective amplifiers. (1,500 to 4,000Hz) At still higher frequencies, one can easily imagine that the lateral energy ratio would show up clearly and with devastating impact in a signal partitioning spiral.

Coupled with realistic sources (i.e., the actual sound system rather than some theoretical omni source), the resultant vectors of magnitude, phase, directivity and resonant amplification can be viewed in all its complexity engendering new insights as to why measurements, to date, have not correlated with subjective listening.

Time Domain Measurements

So far we have discussed our two signals in the frequency domain. What might the correlation and the differences in the time domain look like? More than likely we'd be able to tell the listener's directional orientation to the source merely by looking at the spiral. The time domain display would reveal the size time window for either the inclusion or exclusion of the signal at one ear or both ears.

We need such tools. Redoing the old is nonproductive. Learning new ways tempered by practical experience is the most productive thing any of us can hope to do. The viewer becomes a signal processor capable of detecting directivity via phase information.

As if this weren't enough, the gifted listener has significant differences between his or her ears and they are connected to separate sides of the brain which allows both correlation and differences to be detected in signals arriving at both ears simultaneously.

The TEF-20 lets us consider displaying this information via the Heyser Spiral approach.

The Ma Factor

The Ma, Mc and N are factors we have added to the Hopkins-Stryker equation since the inception of Syn-Aud-Con.

The architectural modifier Ma is defined as:

$$M_a = \left(\frac{1 - \bar{a}}{1 - a_c}\right)\left(\frac{Q_{act.}}{Q_{theor.}}\right)$$

Most of you are familiar with the limiting case where a perfect loudspeaker *exactly* covers a totally absorptive audience and consequently no reverberation can occur no matter how bad the remainder of the space may be.

The terms $1 - \bar{a}$ and $1 - a_c$ are the reflectivity of the room as a whole and the reflectivity of the audience area alone. Because reflectivity is an acoustic power function we then use the actual Q of the device divided by what the theoretical Q of a perfect device would be that exactly covered the audience area. In real life you are lucky to achieve a $M_a = 2.0$ but that's worth 3 dB in lowered reverberant level.

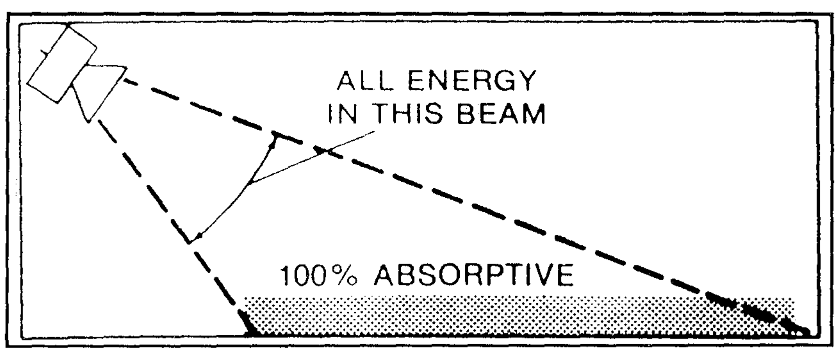
We always try to take full advantage of Ma by getting absorption into the audience area first and taking extreme care to choose and aim highly directional devices at that absorption.

In our Syn-Aud-Con classes the example we give is:

1. Room $RT_{60} = 32$ secs
   (a total impossibility as air absorption would limit the maximum allowed RT_{60} to less than this)
2. 100% absorptive $a_c = 1.0$
   (This also is an impossibility as the typical $\bar{a}$ for a listener is 0.75 - a reflection from this amount down 6 dB)
3. A perfect loudspeaker: covers only the audience and *nothing* else

Therefore, the Ma factor becomes

$$M_a = \frac{(1 - 0.0001)}{(1 - 1)}\left(\frac{100}{100}\right) = \infty$$



Meaningful Auditorium Measurements

A comment is made in a recent article on Architectural Acoustics Research Today, "It is disappointing that so little of this work is being actively pursued in the United States". "This work" is computer modelling, concert hall measurements of the ratio-of-direct-to-reverberant levels, and noise measurements (especially at lower frequencies.)

A careful look at the references reveals that the only work under consideration is in what the writer considers a peer journal . The writer has no "input port" for work done by the peasants out earning a living.

EASE and AcoustaCADD completely outperform any of the programs he references both theoretically and in practical use. None of the referenced programs auralize.

All the concert hall measurements are done with dual channel FFTs and conventional microphones. Finally, the noise criteria data shown reveals the usual intercontinental NIH factors and the pursuit of fleas on the back of fleas. (a second order flea, that is!)

The article reminds me that those of us who are doing the work need to turn completely away from the type of thinking this article expresses and get on with the new and better ways of looking at what we do.

Bats in the Bank

by

Ernie Pence

We recently held an Advanced Acoustic Measurements Workshop at the Medora High School Gym. I went to the local bank in this small S. Indiana town to cash a check. As I approached the teller with check in hand, four tellers ran, with a sudden burst of speed, into a room and barricaded themselves behind a glass enclosure. Not knowing what had happened, I looked behind me to see who or what had entered the bank to cause this action. All money was abandoned, cash drawers were open, leaving me alone in the bank. Suddenly a bat dropped from the ceiling and began his flight to escape. It was at this point that I recognized that the bank was not under any form of attack and that I was perfectly safe.

With the use of a calendar pad and some other financial paper materials I was able to subdue the bat and make the bank safe again for the day's transactions.

We live in a world in which we feel that big is better. But in this small Indiana town and this small bank it is my feeling that this certainly is not true. Here is a bank that refuses to be big; it is here for one purpose and that is to serve the community in which it operates. This is truly a bank in which you enter and the President, who has been there since the bank opened (not as President as he started as a teller), calls you by name. It is a bank in which you are not a number; in fact, my account number has only five digits and the first one is 0 - and I am one of the newer accounts.

The employees of the bank may be afraid of bats, as in this case, but they are not afraid to stand up to the large conglomerates that are swallowing up the small hometown banks. It is this business spirit that we admire.

Two Eared TEF Measurements

The Heyser Spiral is composed of a real part and an imaginary part and depicts the partitioning of the signal between these two parts. The real part normally depicts the amplitude of that part of the signal whose output is in phase with the systems input. The imaginary part depicts that part of the signal whose output is ninety degrees out of phase with the system's input. The imaginary part is a Hilbert transform of the real part. (There's nothing imaginary about what is labelled imaginary - it's just an ancient term that's hung on from the earlier days of mathematics.)

The beauty of the Heyser Spiral is the viewing of the partitioning between the two parts both in the time domain and in the frequency domain.

PARTITIONING

Farrel Becker was the first to apply the Spiral in impedance. He showed that impedance can be looked at in two parts - resistance and reactance. The impedance becomes the amplitude of the spiral.

The left ear as the real part and the right ear as the imaginary. The differences between the two ears provide partitioning. Left and right ears, either as a product (multiplication) or as a sum (addition) of both ears can be depicted as the real part and the difference between them is shown as the imaginary part.

INPUTS

Inputs for these measurements could be "In-the-Ear (ITE) microphones," SASS, or the Aachen Head, to name some first choices.

A BETA TEST PROGRAM

To allow Syn-Aud-Con to look at both the extended feasibility and the utility of such measurements, Ron Bennett of Techron has provided us with a software approach that allows all of the above mentioned measurements.

Because the TEF is not presently a two-channel analyzer we sweep for each ear and store data. This data is then post processed as desired from the labelled data.

IMPORTANT FOR THE FUTURE

We believe that this approach can show us new ways to see what we hear and to see it in a way that more closely relates to our subjective impression of the event or device.

New Edition of "Handbook for Sound Engineers"

The latest edition of the "Handbook for Sound Engineers", edited by Glen M. Ballou is a significant improvement over the earlier edition. The chapters in this second edition are specially written just for this handbook unlike other offerings made up of various technical journal articles.

This volume is startlingly up to date including mention of auralization and the Heyser spiral. Mahlon Burkhard has contributed an outstanding chapter on filters and equalizers that is must reading for any serious audio engineer. Chris Foreman has done an exceptionally thorough job on the Sound System Design chapter. Chips Davis, in one of his rare in-print discussions, has written the chapter on "Details of the Session" co-written with Linda Jacobson. His chapter includes session comments by George Massenburg, Allen Sidess, Tom Jung, Bell Porter and Ron Estes. Full layouts of their microphone setups are included (this alone is worth the price of the book.)

F. Alton Everest has done his usual thorough update on Acoustics including the latest on diffusors, though this subject deserves a full chapter written by Peter D'Antonio. Cliff Hendricksen, Ken Pohlmann, Dale Manquen, Rolly Brook, Ted Uzzle, Gene Patronis (on his other exceptional talent, amplifier design, that is not as well known as his loudspeaker designs), David Miles Huber, George Alexandrovich and Emil Torick, Steve Dove, Franklin J. Miller and Don & Carolyn Davis complete the author's list. Glen Ballou, Al Grundy and Ronald G. Ajemian did a superb chapter on "Transmission Techniques".

This is not a trivial book at 1505 pages and $99.95 U. S. price ($139.95 Canada). It is up-to-date, accurately written information, and the authors mentioned are people we know and respect personally. We regard it as a superb reference and a book you can pick up and read a chapter for a broader view of the industry you live in.

Sound Fields —————— *How They Develop*

Let's talk about large room acoustic sound fields. They can be grossly divided into:

1. the direct sound field
2. the early reflections
3. the reverberant sound field

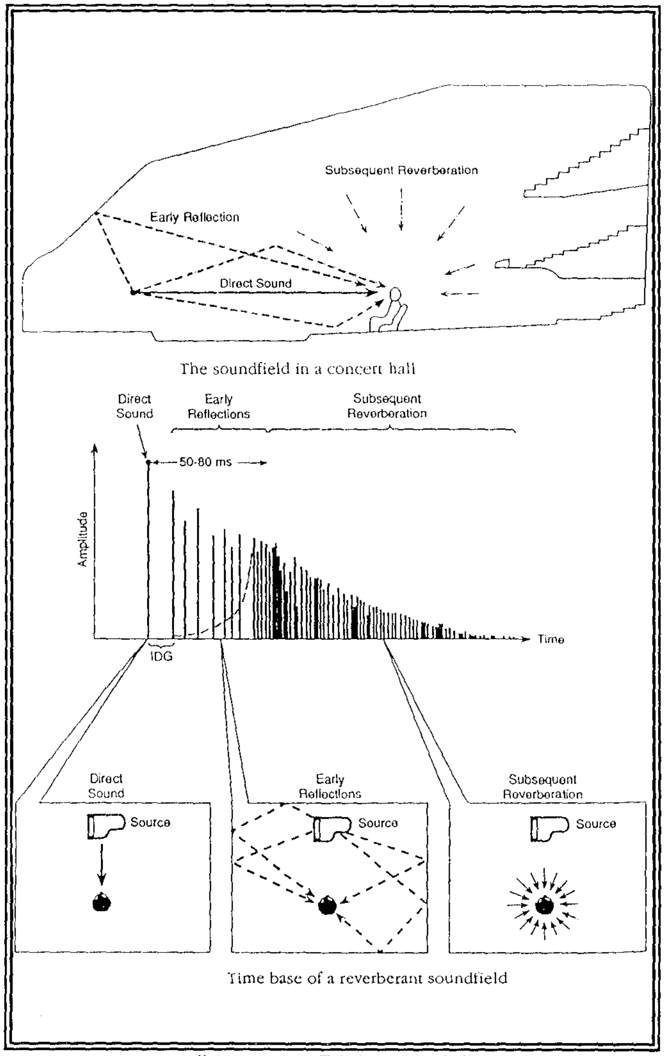
DIRECT SOUND FIELD CHARACTERISTICS

The direct sound takes the quickest path from the sound source to the listener. This is not always a straight line but it's always the quickest path via the given medium. The direct sound follows inverse square law level change plus air absorption effects. The direct sound normally carries the speech intelligibility energy.

EARLY REFLECTIONS

After an initial signal delay gap the early reflections begin to arrive. The early reflections that arrive with levels equal to or less in level than the direct sound and with signal delays from 15 to 50 milliseconds normally do not materially aid or interfere with speech intelligibility.

An early reflection higher in level than the direct sound and with a useable frequency spectrum from 300 to 3000Hz can on occasion be used by the ear/brain system to establish speech intelligibility. When this happens, the direct sound then becomes a form of interference.



The soundfield in a concert hall

Time base of a reverberant soundfield

"Advanced Digital Audio"
Jayant Datta (Author) and Sams (Publisher).

The early reflections do play several very important roles:

1. They are, in conjunction with the direct sound, additional loudness
2. They determine the acoustic perception of a hall (i.e., how live, how large, how diffuse) by the level of return, the delay of return and the distribution of return.

THE REVERBERANT SOUND FIELD

The reverberant sound field is the classic acoustic measurement. While early workers concentrated on its time of decay after the cessation of a stimulus (RT_{60}) it has been found that its initial level is of greater importance than its time of decay. A reverberant sound field that has an RT_{60} of 10 secs but is only 3 dB above ambient noise level is barely audible. An RT_{60} of 3 secs in a space where the reverberant level is as high as speech is devastating.

DIRECTIONAL PROPERTIES OF ALL THREE FIELDS

The direct sound is directional in the sense that it is easy to hear where the source is when direct sound predominates. Early reflections are usually not too specific in direction unless they are high enough to also interfere. Reverberation is an immersion in sound from all directions and negates any directional properties of either microphones or loudspeakers.

These simple generalities and the accompanying illustrations should serve to provide an intuitive feel for these major categories.

New Shure Booklet: Microphone Selection and Applications for Church Sound Systems

Tim Vear, a Shure applications engineer, has written a booklet entitled *Microphone Selection And Application For Church Sound Systems*. This 36 page booklet does a splendid job of putting the essentials into lay language without sacrificing technical integrity or utilizing half truths. For example, the relative ineffectiveness of directional microphones in increasing acoustic gain is covered with "In fact, the potential acoustic gain (maximum gain before feedback) is almost completely dependent on the relative distances between the sound source, the microphone, and the loudspeaker ". . . "This maximum gain is only partially dependent on the directionality of the microphone and loudspeaker." I can easily imagine the iron discipline it takes on the part of a microphone manufacturer to not simply use the old myth "Directional microphones provide higher acoustic gain." Tim treats the subject of "reach" with the same uncompromising integrity.

If I were a sound contractor, I would include this booklet in every church submittal I generated. This is a well organized, carefully prepared, and professionally presented piece of literature. Reading Tim Vears' biography reveals that he has indeed "paid his dues" in the audio industry with both formal degrees and extensive practical experience.

Incidentally polarity is called polarity in this booklet. As you surely can tell by now, we are impressed with this excellent booklet. The hardest writing jobs there are spring from trying to maintain technical integrity while writing in simple layman language. Tim Vears' effort is a totally successful achievement.

The Difference Between Sound Pressure, Sound Intensity, & Sound Power

To begin with, it would be useful to first look at the base units for sound pressure, P, power per unit of area, I, and total power, W.

Pressure

Pressure is defined in base units as:

$$\frac{\mathrm{Kg}}{\mathrm{M}\cdot\mathrm{S}^2} \qquad \left(\mathrm{KG}\cdot\mathrm{M}^{-1}\cdot\mathrm{S}^{-2}\right)$$

That means that one kilogram of mass per meter (per second per second) represents the pressure, P, that appears at a point of measurement. Since energy is defined in base units as:

$$\frac{\mathrm{M}^2\cdot\mathrm{Kg}}{\mathrm{S}^2} \qquad \left(\mathrm{M}^2\cdot\mathrm{KG}\cdot\mathrm{S}^{-2}\right)$$

and is named Joule, we can then find a Joule per second which defines power named watt. Therefore, power has base units of:

$$\frac{\mathrm{Joule}}{\mathrm{sec}} = \frac{\mathrm{M}^2\cdot\mathrm{Kg}}{\frac{\mathrm{S}^2}{\mathrm{S}}} = \frac{\mathrm{M}^2\cdot\mathrm{KG}}{\mathrm{S}^3} \quad \left(\mathrm{M}^2\cdot\mathrm{KG}\cdot\mathrm{S}^{-3}\right)$$

Note that I (intensity in $\mathrm{W/M}^2$) is power per unit of area. (Per means divided by.) The power is divided over an area.

When all areas through which power is flowing are accounted for, we then obtain the total power, W.

The sound pressure is analogous to voltage across a given component in a circuit. The sound intensity is analogous to the power consumption of a given component in a circuit (i.e., so many watts are dissipated in that component).

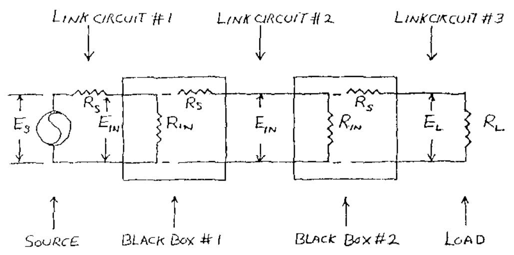
Finally, the total sound power, W, is analogous to the total electrical power, drawn by the entire circuit, from the wall plug. Like voltages, sound pressures when squared are proportional to sound power, hence the use of the 20 multiplier with voltages and sound pressures when they are converted into levels.

It is always helpful to realize that the human receiver responds to variations in sound pressure and that any and all attempts to supply the required sound pressure for the least expenditure of sound power is called sound engineering.

The use of increased Q's in electroacoustic transducers, shortening of the distance between the transducer and the listener, Dx, and controlling the excess power used via acoustic absorption, diffusion, and geometry are all legitimate engineering tools available to knowledgeable persons.

The Black Box Approach

BLACK BOX TERMS AND EQUATIONS



$$L_{AIP} = 10 \, LOG\left[\frac{E_S^2}{0.001 R_S}\right] - 6.02 dB$$

$$L_{OUT} = 10 \, LOG\left[\frac{E_L^2}{0.001 R_L}\right]$$

$$SYSTEM \; GAIN = 20 LOG\left(\frac{E_L}{E_{IN}}\right) + 20 LOG\left(\frac{R_{IN}}{R_S + R_{IN}}\right) + 10 LOG\left(\frac{R_S}{R_L}\right) + 6.02 dB$$

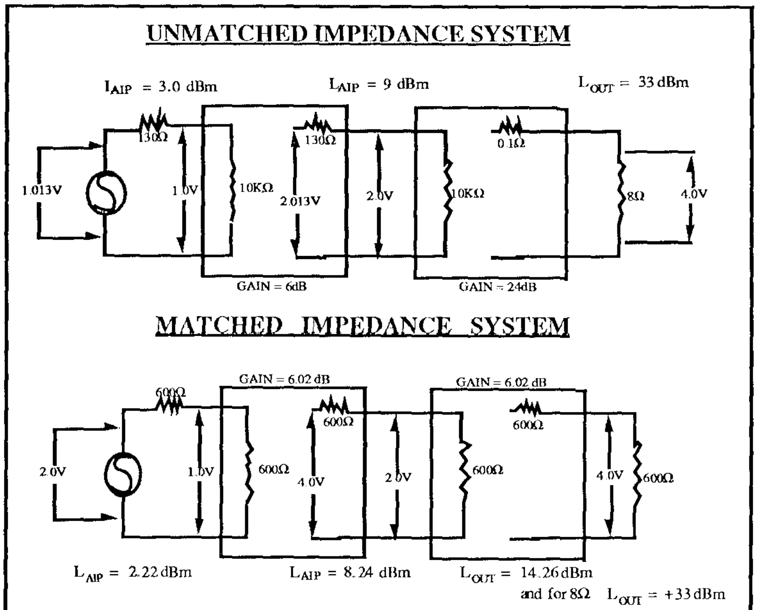
$$BLACK \; BOX \; GAIN = \left[10 LOG\left(\frac{E_L^2}{0.001 R_L}\right)\right] - \left[10 LOG\left(\frac{E_S^2}{0.001 R_S}\right) - 6.02 dB\right]$$

There is component engineering (what manufacturers do) and systems engineering (what contractors do). Component engineers work in voltages and currents. Systems engineers work with levels and, at the output, with audio power. Components used in systems can be treated as "black boxes" or devices under test, DUT) where the only concern need be the input and output ports (a two-port device) or the input only (a single port device - a voltmeter, for example).

If we are to engineer the system as distinquished from merely assembling a series of components, we need to determine the following:

1. the input and output *levels*(Voltages *are not* levels.)
2. the input and output impedances
3. the input and output polarities
4. the input and output circuit configurations (balanced-unbalanced)
5. the type of connectors required, grounding and shielding needed, and current carrying capacities required for the cables.

All of the above needs to be gathered and evaluated.



BLACK BOX NOMENCLATURE

The terms most commonly encountered are shown in Figure 1. To ascertain the gain or loss of the Black Box #1, we need to find the LAIP for the source followed by LAIP of Black Box #1 as a source to Black Box #2. The LAIP of the source is then subtracted from the LAIP of Black Box #1 as a source to Black Box #2.

Again, using the LAIP of Black Box #1 as a source to Black Box #2, subtract it from the LOUT of Black Box #2 to obtain the gain or loss of Black Box #2.

The total system gain should, of course, be equal to the sum of the individual Black Box gains.

"Joe Took Father's Shoe Bench Out"

"Joe took father's shoe bench out."

"This nonsensical sentence is chosen because it and its mate, 'She was waiting at my lawn.' contain all the fundamental sounds in the English language that contribute appreciably towards the loudness of speech."

The quote above is from Harvey Fletcher's *Speech and Hearing in Communications*, a true classic text by any standard.

We often add: "...for a canoe ride down the Tippecanoe river in a twin screw stainless steel cruiser." to "Joe took father's shoe bench out," but it isn't part of the original sentence.

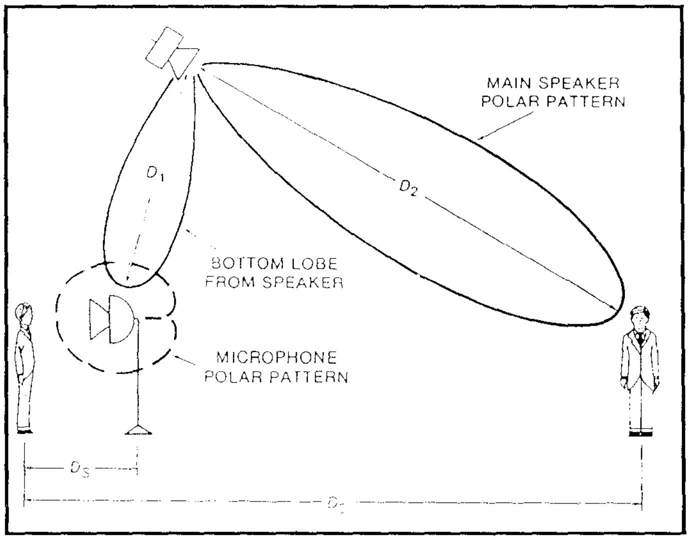
The Me Factor

The letters M_e stand for electroacoustic modifier. It can be invoked by a listener cupping his or her ears toward the sound source. It is more commonly encountered in the form shown in. Fig. 1 The optimum case would be outdoors with no reflective surfaces present.

When you have a directional loudspeaker and a directional microphone and both are in each others free field (no reflections), then the potential increase in acoustic gain can be obtained by orienting the polar responses of the two devices so as to effectively increase D_1's acoustic distance. For example, if an omni pair of devices would feed back at 70 dB and they were replaced by a microphone 3 dB less sensitive at the angle toward the loudspeaker than it is on axis plus the loudspeaker's output at the angle toward the microphone was 3 dB lower than the on axis signal, then a potential 6 dB of increased acoustic gain is possible.

Be aware that all directional devices will be omni at some frequencies and M_e calculations are only valid if those susceptible frequencies are not used.

Here at Syn-Aud-Con we normally leave M_e at 1.0 for our initial calculations. For outdoor systems and for small dead rooms (i.e., conference rooms) it can prove to be of use.



The M_e Factor

The electroacoustic modifier M_e is defined as:

$$M_e = 10\left(\frac{L_{M \Rightarrow T} - L_{M \Rightarrow L/S} + L_{L/S \Rightarrow L} - L_{L/S \Rightarrow M} +}{20}\right)$$

Where: $L_{M \rightarrow T}$ is the relative change in sensitivity between on axis and the angle from microphone to talker.

$L_{M \rightarrow L/S}$ is the relative change in sensitivity between on axis and the angle from microphone to loudspeaker.

$L_{L/S \rightarrow L}$ is the relative change in level between on axis and the angle from loudspeaker to listener.

$L_{L/S \rightarrow M}$ is the relative change in level between on axis and the angle from loudspeaker to microphone.

A Basic Change at Syn-Aud-Con

This summer Carolyn and I came to the decision that we would no longer present "on-the-road" classes. All Syn-Aud-Con "basics" classes will hereafter be held at "The Farm." The Farm classes are three days and include "hands on" opportunities. Three days allows a more detailed approach to the subjects discussed and having equipment permanently set up allows far more sophisticated demonstrations especially of those computer based measurement and design programs that all of us need to know about.

Not travelling will eventually result in more writing and research by us and better and better tools to put in your hands.

Since Syn-Aud-Con advertises only sporadically, we rely on our graduates to help us by "word of mouth" to reach those needing the Syn-Aud-Con touch.

It's also fitting to express our deep gratitude to our 8000 grads over the past nineteen years for your attendance at our "on-the-road" classes. Our newsletters and workshops are a partial expression of our respect and appreciation for you.

Speaking to those of you who haven't been through a Syn-Aud-Con class in the last three years, what's going on now in our classes is even more exciting than the early TEF days with the TEF 20, auralization, and the explosion of digital technology we hoped for twenty years ago and now have fully upon us.

So Syn-Aud-Con is continuing to help the newcomers, support the grads, and inspire the super achievers among you. We sincerely hope a majority of you get the chance to experience the Farm and see for yourselves that Syn-Aud-Con is proceeding full speed ahead into the twenty first century.

A Professional Interface –Audio Technologies Inc.

Audio Technologies Inc, ATI, 328 W Maple Ave, Horsham, PA 19044, PH 215-443-0330 manufactures a pair of boxes that belong in the tool kit of any engineer trying to interface semi pro and consumer audio equipment with professional devices. We have been using these in our farm classes as a part of our exercises on interfacing incompatible devices.

The clarity of their product labeling, the straightforwardness of their warranty, and the thoroughness of their specifications, along with a performance that meets them, demands respect for their offering.

SIMPLE LIMITED WARRANTY
ATI warrants that:

Your Interface will work when you get it.
Your Interface will do what our published specs say it will do.
Your Interface will continue to do the above for at least one year.

As Long As:

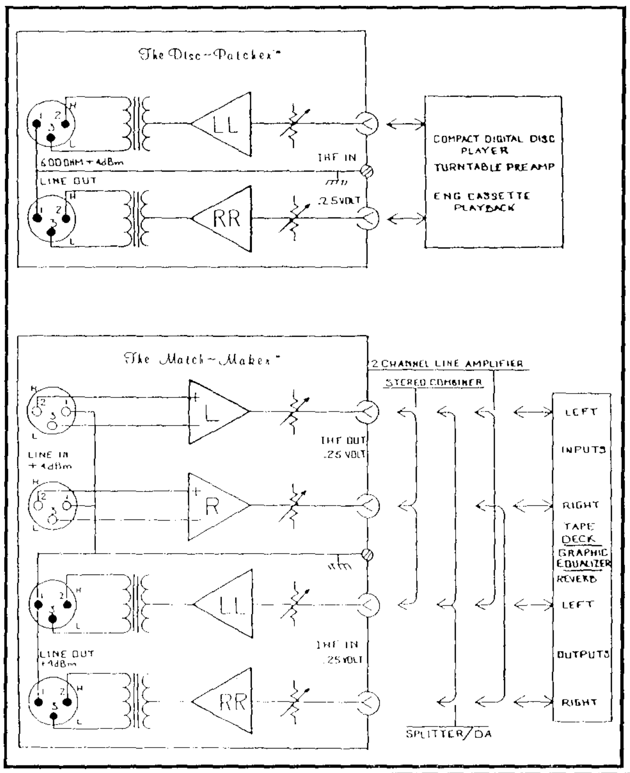
You don't use it as an anvil.
You don't rip out the audio connectors.
Your power company treats it right.
You don't adjust the pots with a crowbar.
You don't take it swimming

If it doesn't work, call us first.
We will immediately:

Tell you with a straight face that you are the first person who ever had a problem with one of our Interfaces.
Send you a replacement part or
Send you a replacement unit.
Ask you to return the defective unit prepaid
Help you put a damage claim to the shipper
Recommend you to a competitor.

We are not responsible for:

Acts of God
Murphy's Law
The wrath of your boss and other consequential damage



As the excerpts from their instruction manual make clear, these men have enough engineering skill to have regained their sense of humor after facing the "real world".

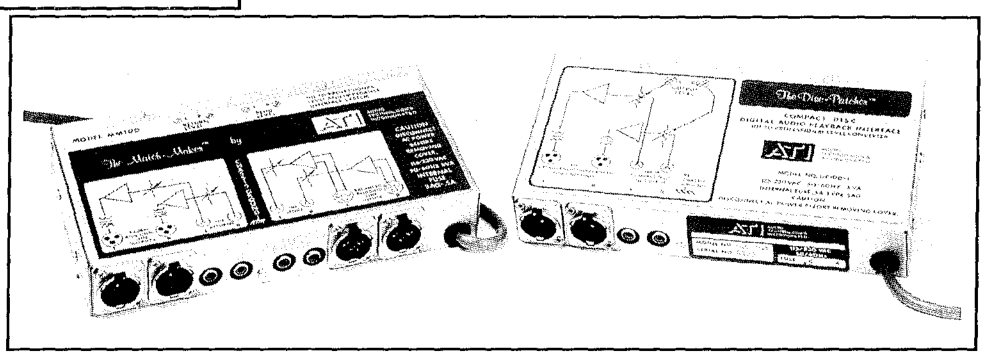
If you call Syn-Aud-Con with an interface problem between incompatible equipment we will refer you to this article for the solution.

Excerpts from **APPLICATIONS**

The Disc-Patcher™ and the Match-Maker™ are level and impedance matching interfaces for semi-pro, industrial and consumer audio equipment operating into professional balanced 600Ω systems.

The Disc Patcher is a uni-directional stereo interface for Playback Only applications. ....The Match-Maker is a bi-directional interface which bridges a stereo pair of 600Ω balanced or unbalanced, +4dBm lines and converts those signals to a nominal .25 Volt (-10dBu) level to fee, for example, the record inputs of a consumer cassette or reel-to-reel tape recorder.......The Match-Maker IHF connectors may also be jumpered for use as a two channel 600Ω Line Amplifier, a two output Distribution Amplifier or a Mono Summing Amplifier.

In our opinion, these are the boxes you will need in an emergency situation and that should be on hand in the same case you carry your cable tester, GLIT, and polarity checker. Yes! people do try to operate without these tools, but the question is WHY?



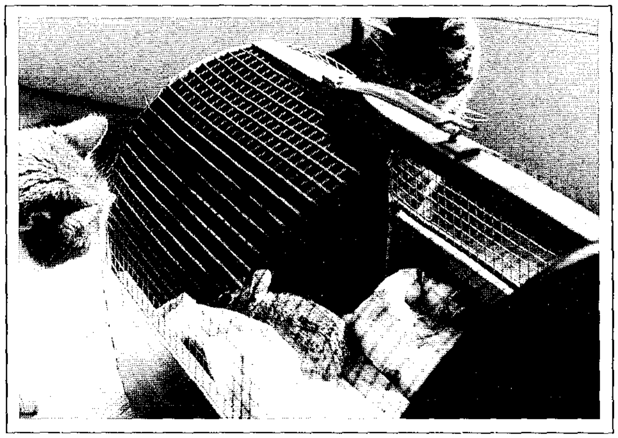
A Rabbit and Two Friends

About mid-April we found a little baby rabbit in our dog's mouth. We rescued it, but it was cold and almost stiff. It was only a few inches long. We warmed it on the stove, put warm pet milk on its lips until he began to suck it in. We added baby-food carrots, later grated carrots and then rabbit food.

We named him Pascal, but we quickly changed his name to PooPoo as he had a very efficient digestive system. PooPoo grew sufficiently so we moved him from the cat-carrier to a large dog cage—which we had to cover with chicken wire, except for the top. When PooPoo jumped straight up and through the top of the cage—twice in 15 minutes we knew it was time for PooPoo to have his freedom, so we took him to a neighbors home who had no dogs. Poopoo was seen several times playing in their yard.

Raising wild animals can have its pain. We watched a crippled baby bird with a damaged wing for a few days in the barn and decided that he was getting no food, so we brought him to the house and put him in PooPoo's cage. We thought he was a crow and named him Edgar Allan Crow, but it turned out that he was a starling - not a bird that is looked upon with favor on the neighboring farms, but we enjoyed him. We fed him every couple of hours on a diet of hamburger and then hamburger rolled in thistle bird seed. Often Edgar would eat perched on my hand with the three cats and two dogs circled around us.

Edgar was with us for about two weeks when I made the mistake of leaving Edgar in his cage outside one night. The next morning there was nothing but feathers. Some wild creature had ripped off the chicken wire and reached in for Edgar. Neighbors thought probably it was a mink or weasel. The next day the Sunday paper carried an article about Mozart's pet starling, quoting an article from <u>American Scientist</u>. It didn't ease the pain any to realize that Mozart loved his starling also.



Our baby rabbit, PooPoo, with his two friends, Pete and Tilly.

Power & Grounding Drawings

David H. Kaye, Consultant in Acoustics in Arlington, MA has graciously allowed us to share his current power and grounding detail drawings.

The following excerpts are from a letter from David Kaye:

"Enclosed are copies of 'my' current power and grounding detail drawings. ("My" in quotes because, although these particular versions are mine, they depend upon suggestions and previous versions from others.)

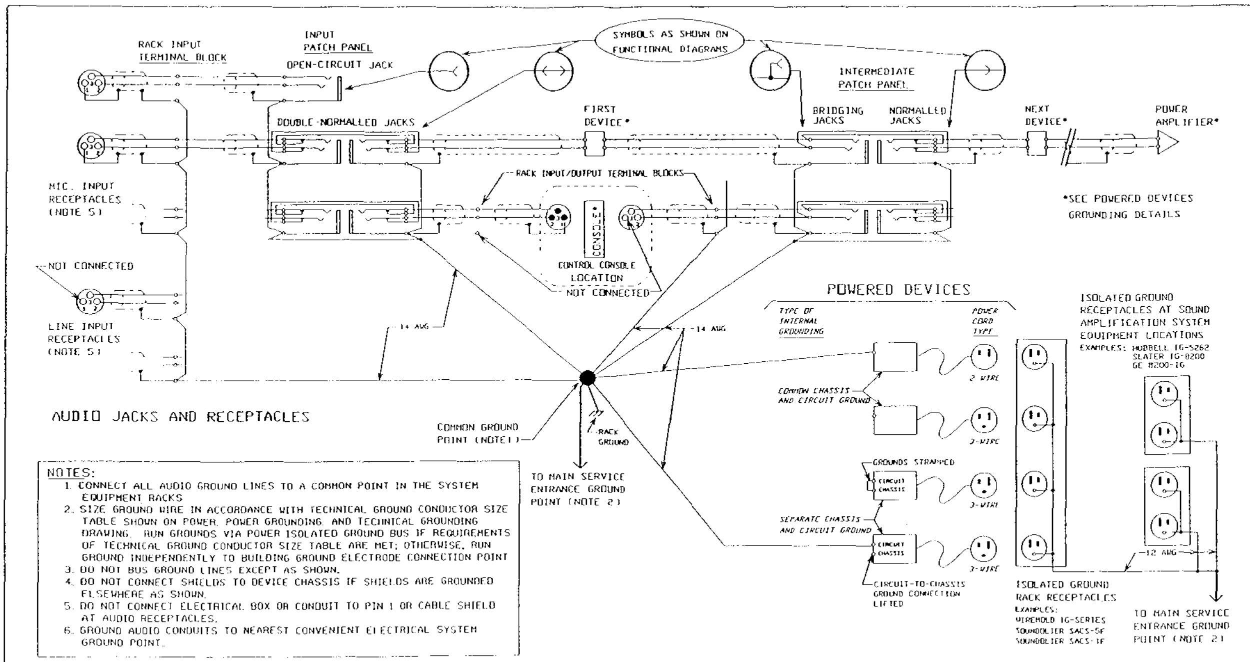
"If it suits your purposes, I have no objection to your sharing these drawings. If the drawings are directly copied, I should receive credit; if the drawings merely become the basis for someone else's drawing, credit is not necessary. I have borrowed heavily from others and I have no objection to folks borrowing in turn from me.

"My grounding wire sizes are based upon maintaining a ground resistance of 0.5 Ω or less.

"On the subject of grounding, let me pass along my list of NEC '90 references regarding isolated grounding. These are what I have relied upon in preparing my detailed drawings:

Primary descriptions:
250-74, Exception No. 4
250-75, Exception

Additional information
250-74, Fine Print Note
250-91 (c)
384-3 (c)
384-20, Exception"



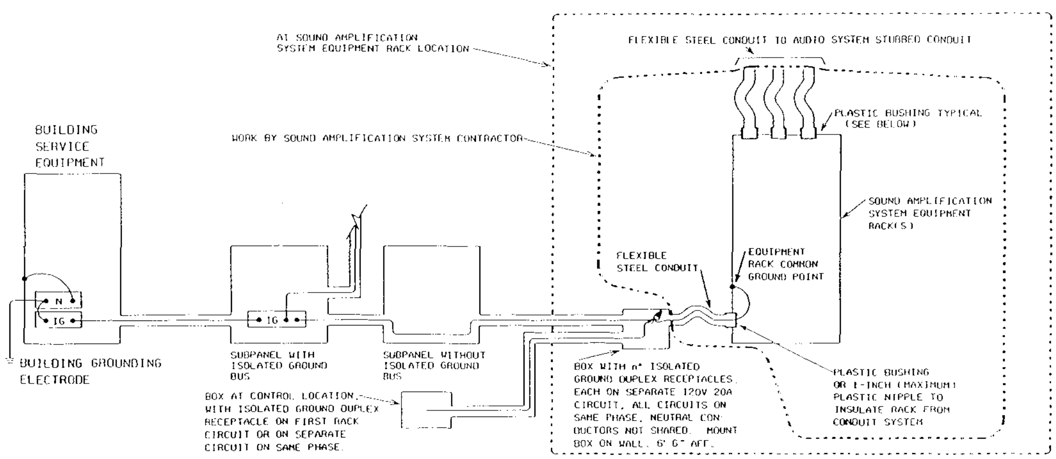
NOTES:
1. CONNECT ALL AUDIO GROUND LINES TO A COMMON POINT IN THE SYSTEM EQUIPMENT RACKS
2. SIZE GROUND WIRE IN ACCORDANCE WITH TECHNICAL GROUND CONDUCTOR SIZE TABLE SHOWN ON POWER, POWER GROUNDING, AND TECHNICAL GROUNDING DRAWING. RUN GROUNDS VIA POWER ISOLATED GROUND BUS IF REQUIREMENTS OF TECHNICAL GROUND CONDUCTOR SIZE TABLE ARE MET; OTHERWISE, RUN GROUND INDEPENDENTLY TO BUILDING GROUND ELECTRODE CONNECTION POINT
3. DO NOT BUS GROUND LINES EXCEPT AS SHOWN.
4. DO NOT CONNECT SHIELDS TO DEVICE CHASSIS IF SHIELDS ARE GROUNDED ELSEWHERE AS SHOWN.
5. DO NOT CONNECT ELECTRICAL BOX OR CONDUIT TO PIN 1 OR CABLE SHIELD AT AUDIO RECEPTACLES.
6. GROUND AUDIO CONDUITS TO NEAREST CONVENIENT ELECTRICAL SYSTEM GROUND POINT.

DAVID H. KAYE, CONSULTANT IN ACOUSTICS
39 BRAND STREET, ARLINGTON, MA 02174
TELEPHONE: 617/643-6430 FACSIMILE: 617/643-6999

SOUND AMPLIFICATION SYSTEM
JACK, RECEPTACLE & DEVICE
WIRING & GROUNDING DETAILS

DATE: 91/09/10
SCALE: NONE
DRAWN BY: GCP

SHEET Nº
SAS Z



NOTES:
1. WORK BY THE SOUND AMPLIFICATION SYSTEM CONTRACTOR SHOWN ON THIS SHEET INCLUDES SOUND AMPLIFICATION SYSTEM EQUIPMENT RACK(S), STEEL FLEXIBLE CONDUIT ATTACHED TO RACK(S) VIA INSULATING BUSHING OR PLASTIC NIPPLE, AND GROUND CONDUCTOR WITHIN FLEXIBLE CONDUIT AND RACKS. OTHERWISE, WORK SHOWN ON THIS SHEET IS BY ELECTRICAL CONTRACTOR.
2. "IG" = ISOLATED GROUND. IG CONDUCTOR SHALL BE INSULATED. IG SYSTEM SHALL HAVE NO CONTACT WHATSOEVER WITH METAL CONDUIT/RACEWAYS, PANELBOARD ENCLOSURES, AND NEUTRAL CONDUCTORS EXCEPT WHERE IG BUS IS BONDED TO NEUTRAL BUS AT BUILDING MAIN SERVICE EQUIPMENT
3. IG CONDUCTOR IS USED AS TECHNICAL GROUND. SIZE PER TECHNICAL GROUND CONDUCTOR TABLE AT RIGHT OR PER NEC-90 TABLE 250-95, WHICHEVER REQUIRES LARGER SIZE.
4. CONDUCTORS AT BUILDING SERVICE EQUIPMENT CONNECTING IG BUS TO NEUTRAL AND NEUTRAL TO BUILDING GROUNDING ELECTRODE SHALL BE AT LEAST SIZE OF LARGEST IG CONDUCTOR.
5. USE STEEL CONDUIT/RACEWAY FOR ALL SOUND AMPLIFICATION SYSTEM POWER, GROUND, CONTROL, AND SIGNAL LINES EXCEPTION: LOUDSPEAKER LINES, FOR WHICH STEEL, OTHER METAL, OR PVC MAY BE USED.
6. WIRING PROVISIONS SHOWN AT SUBPANELS ARE TYPICAL FOR SUBPANELS, IF ANY.
7. POWER PROVISIONS FOR EQUIPMENT RACK(S) ARE SHOWN HERE AS WALL-MOUNTED RECEPTACLES, INTO WHICH RACK POWER CORDS WILL BE PLUGGED. ALTERNATIVELY, POWER MAY BE HARD-WIRED INTO THE RACK(S). IF SO, PROVIDE A RACK POWER DISCONNECTING DEVICE WITHIN THE RACK ROOM.
8. FIRST RACK POWER CIRCUIT IS INTENDED FOR INPUT AND SIGNAL-PROCESSING EQUIPMENT; OTHER CIRCUIT(S) FOR POWER AMPLIFIERS

TECHNICAL GROUND CONDUCTOR SIZE TABLE

DISTANCE FROM EQUIPMENT RACK COMMON GROUND POINT TO BUILDING GROUNDING ELECTRODE CONNECTION POINT	MINIMUM COPPER CONDUCTOR SIZE (AWG)* (INSULATED WIRE)
60 FT OR LESS	12
60 - 95 FT	10
95 - 150 FT	8
150 - 235 FT	6
235 - 385 FT	4
385 - 475 FT	3
475 - 590 FT	2
590 - 770 FT	1
770 - 910 FT	1/0
910 - 1100 FT	2/0
1100 - 1500 FT	3/0
1500 - 1900 FT	4/0

*USE LARGER SIZE IF/AS REQUIRED IN ACCORDANCE WITH NATIONAL ELECTRICAL CODE TABLE 250-95.

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* *QUANTITY OF CIRCUITS AS REQUIRED FOR PROJECT*

SOUND AMPLIFICATION SYSTEM
POWER, POWER GROUND, AND
TECHNICAL GROUND DETAILS

DATE: 91/09/10
SCALE: NONE
DRAWN BY: GCP

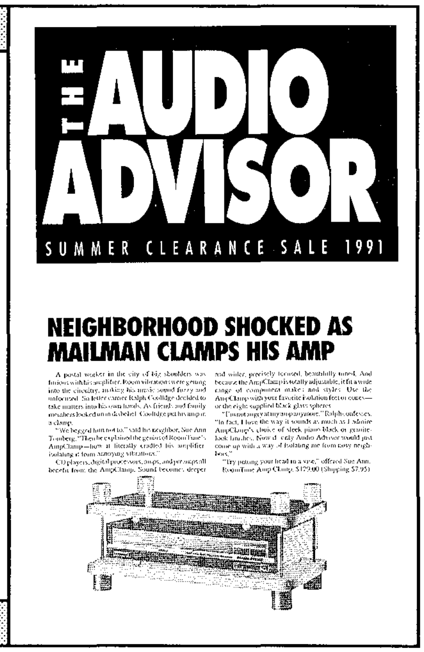
SHEET Nº
SAS X

The Silly Season

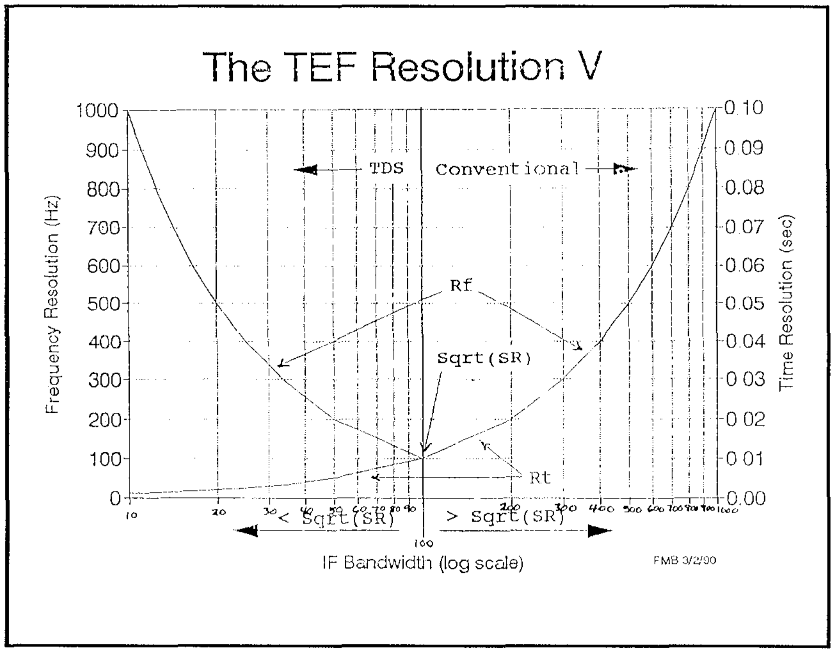
What with "temporal force", "Wire conditioning", why not clamp your electronics? I know, don't tell me, you could hear a difference. Yes, yes, go quietly to the nice man in the white coat and we will not hold it against you.

"Sound becomes deeper and wider, precisely focused, beautifully tuned"

We have tried for a mental image of the above but somehow it comes out as three contradictions in a single sentence - maybe we were not sufficiently artistically sensitive.



Logarithmic & Linear at Two Frs



TEF analysis primarily used linear frequency scaling because it is a constant bandwidth analysis system. It can, of course, display this information logarithmically as well provided that the frequency resolution (f_r) is sufficient to allow accuracy at the lowest frequency of interest on the log scale. Our practice here at Syn-Aud-Con is to measure at the optimum bandwidth ($B_{opt.}$) where:

$$B_{opt} = \sqrt{\text{sweep rate in Hz per sec.}}$$

Then by using (TEF 10 or 12), in the cursor mode, lower case *f* for the filter sub routine we can reduce excessive resolution to any lesser resolution.

PERTINENT TEF PARAMETERS

The equations below serve as an excellent check list to insure the legitimacy of your measurement setups. Farrel Becker's most useful illustration of the TEF resolution V for a sweep rate of 10,000 Hz/sec. provides insight into the relationship between time resolution (labeled R_t on the chart) and frequency resolution (labeled R_f on the chart).

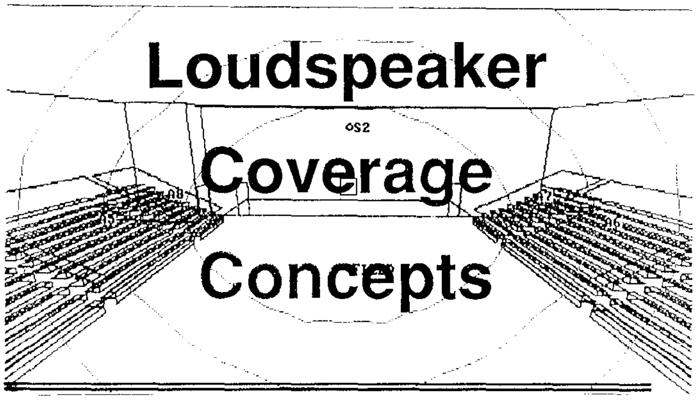
$$R_t = \frac{B}{S}$$

Where if bandwidth "B" is less than the square root of the sweep rate "S".

$$R_f = \frac{S}{B}$$

When the bandwidth is greater than the square root of the sweep rate, then the $R_f = B$ and $R_t \frac{1}{B}$

Loudspeaker Coverage Concepts



How to choose and aim loudspeakers has a scientific basis and is a genuine art in application. Coverage is only one of the triumprate of intelligibility, coverage, acoustic gain. Many computer programs consider only coverage and quite often at the expense of both intelligibility and acoustic gain to the point where the system is judged by the live listener as unsatisfactory.

It is possible to simultaneously solve for coverage and intelligibility and Syn-Aud-Con employs the following method to do so.

First consider what the ideal limiting case would be. A loudspeaker of sufficient Q to insure intelligibility at the remotest listener and perfectly even coverage for all the seats. (See Fig. 1.) What's most interesting about this limiting concept is that if the minimum Q for speech intelligibility at the furthest seat can be met and coverage achieved; it also means that speech intelligibility will be as uniform as the coverage.

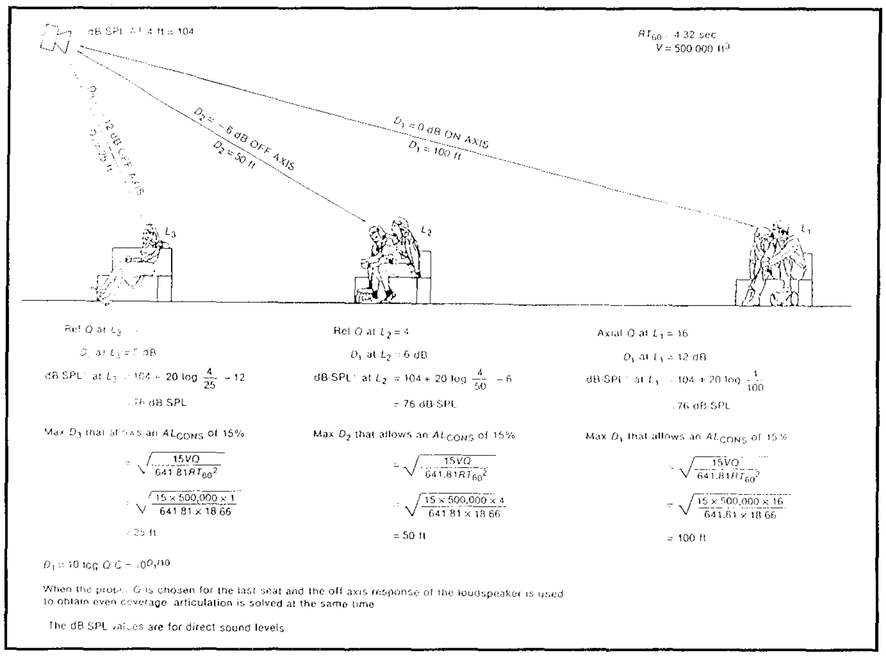
The first step is to compute the required minimum Q for the speech intelligibility score acceptable in your design.

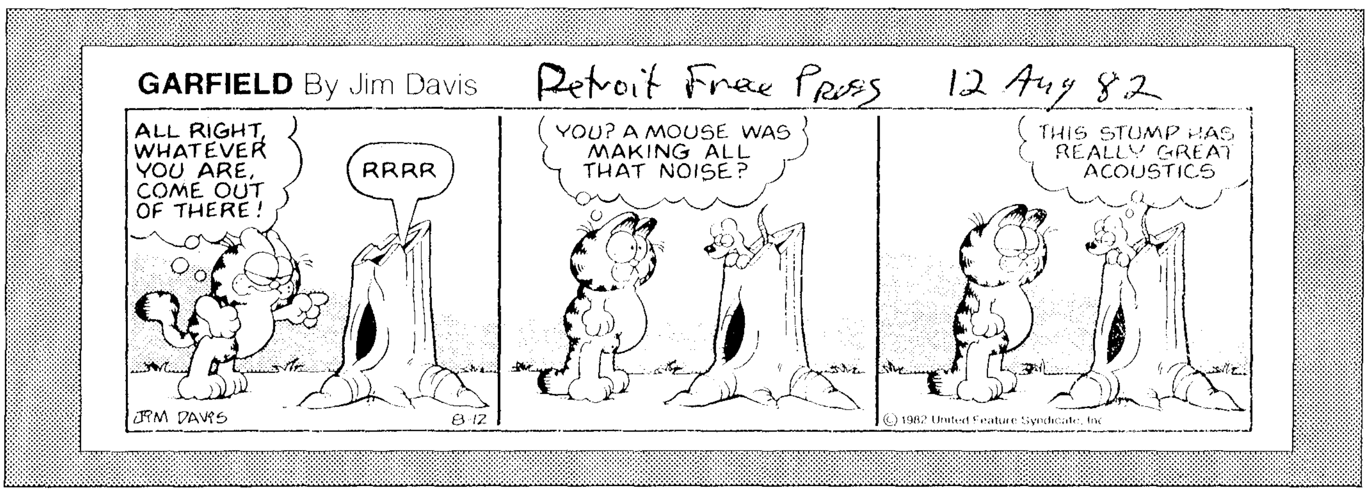
$$Q_{MIN} = \frac{656\,(D_2)^2\,(RT_{60})^2}{15V}$$

The second step is to check coverage. In the ideal case (See Fig. 1) half way back to the source the listener will find that they are at the -6dB coverage angle and for a listener three quarters of the way back to the source they are at the -12dB coverage angle.

Note carefully that the apparent Q at the listener (i.e., the ratio of direct to reverberant sound) is changing constantly as you move further and further off axis, but that this causes no trouble as the distance from the source is changing as well. The equations under the illustration demonstrate this ideal case.

Amateurs try to achieve coverage by varying the levels to multiple speakers in reverberant spaces. This approach is legitimate outdoors or in very dead spaces where all the listeners are at a distance from the source that is shorter than critical distance D_C. It is not correct when in reverberant spaces inasmuch as the loudspeaker with the highest sound power level sets the level of the entire reverberant field. Coverage in such spaces needs to be accomplished by choosing sources with applicable Q values for the distances from its on axis listeners.





Brain Sees Vowels Differently

Associated Press

NEW YORK—*The distinction between vowels and consonants may be more than a classroom lesson, says a study suggesting that the brain handles the two classes of letters differently.*

Two men who had suffered strokes had particular trouble with vowels when they tried to spell words, a researcher said. Their difficulties are described in today's issue of the journal Nature by Roberto Cubelli of the Ospedale Maggiore in Bologna, Italy.

Strokes disable specific parts of the brain by interrupting the blood supply.

In one case, a 43 year old engineer omitted all vowels when he wrote his name and the names of his home town and five objects. He left blank spaces between the correctly written consonants and, though aware of the omissions, did not seem able to choose the correct vowels, Cubelli said. He said this impairment was "completely novel" in medical science.

In the other case, a 62 year old retired typographer made significantly more errors mostly substitutions or transpositions with vowels than consonants when he wrote or recited spellings.

In a Nature editorial, John Marshall of the Radcliffe Infirmary in Oxford, England, said the cases provided a new and unexpected form of writing deficit.

It is not clear why the brain would be sensitive to the distinction between vowels and consonants, Marshall said. It is also hard to imagine that other patents could write vowels but not consonants, he said.

Jay Paul of Pineland Communications Systems in Atlantic City, NJ sent us the Associated Press release from the Philadelphia Inquirer. It is more evidence of the multichannel nature of the brain's signal processing. We haven't seen the details of this study but we'd guess that consonants are "left hemisphere" and vowels are "right hemisphere".

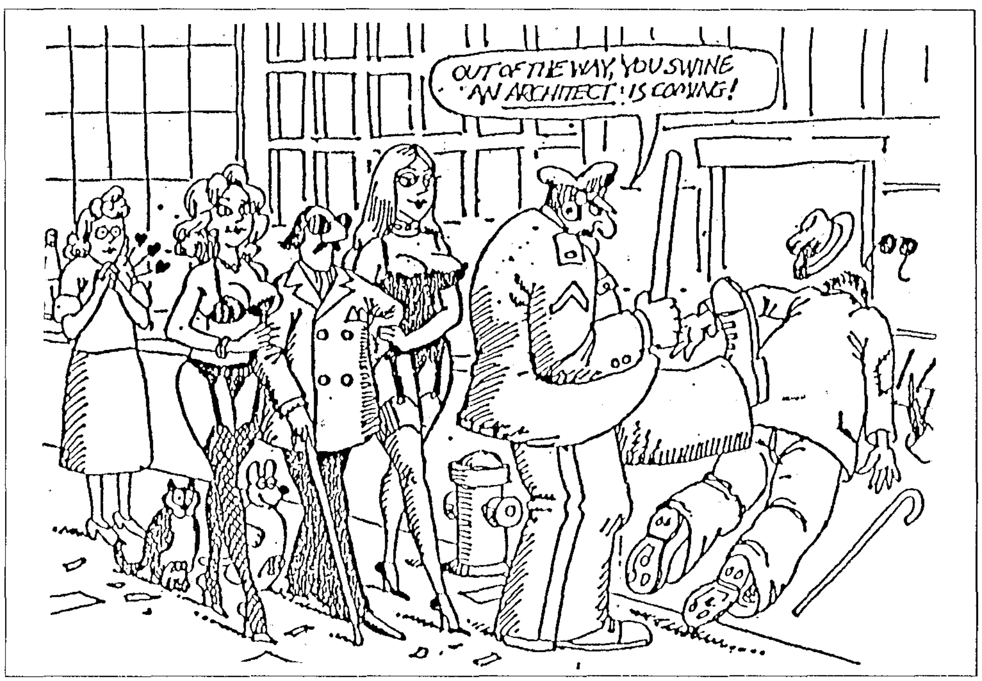
The fascinating question is, "Do trained individuals use different places in the brain for their skills?" (seeing that they have to be learned), or do they instead enhance some part of the brain that otherwise wouldn't utilize its full potential.

The very large number of persons working on these kind of projects should make the 21st century a fascinating experience.

Smile Architect

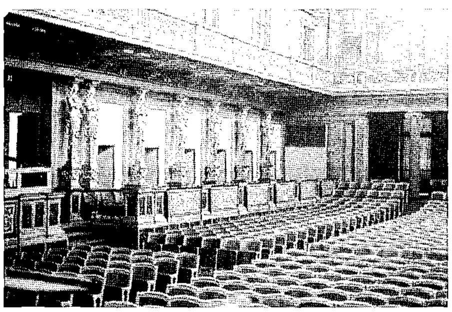
"I'll make it so beautiful you won't care how it sounds!"

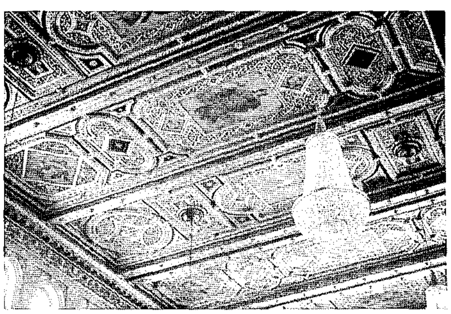
This is my favorite true quote from an architect, which illustrates the powerful ego of some architects.



Diffusion 1979

We're patting ourselves on the back. We were browsing through some old issues of the Newsletters looking for some pictures when we spotted the following article on diffusion in the April 1979 issue.



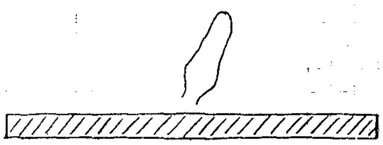


Side and ceiling view of the Musikverienssaal—Vienna showing natural diffusion

DIFFUSION

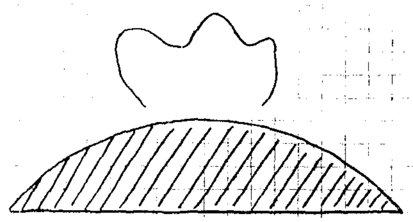
Pick up any book with pretensions to knowledge about recording studios and almost without exception the material on the internal acoustics exhibits an enormous void of accurate or useful information. Implied is that all you have to do is add absorption, with the aid of some devil's apprentice with info from the dark domain, and all is well.

TDS has clearly shown us that the absorption part is easy. Lots of it and all in front of the mixer. The tough part is obtaining the *optimum* diffusion from the "live" end. In fact, the difference in quality of control rooms is the difference in diffusion present at the mixer's ears. The more diffuse and mixed the total sound field at the mixer's ears the better the quality of the sound.



1a. Flat Surface Reflection

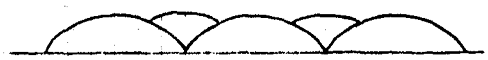
Looking at Illustration # 1 we can see that flat surfaces are not as good as *properly* curved surfaces.



1b. Cylindrical Surface Reflection

Illustration # 2 shows the various techniques used in studios and control rooms. Any combination of these may be used, as well.

2a. Splays



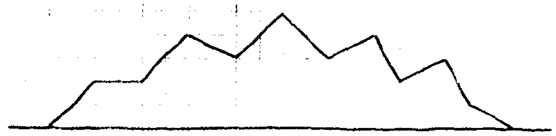
2b. Polycylinders

2c. Mixed Randomly Placed Absorptive & Reflective Surfaces

Manfred Schroeder has proposed a fundamental way of modifying flat surfaces into splendid diffusers.

Illustrations #3 & 4 show his technique for one plane. Obviously both planes should be accounted for in actual application.

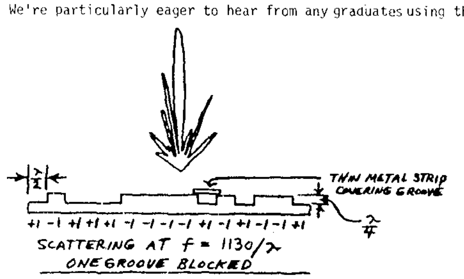
Four or five different panels designed for center frequencies, say of 500, 1000, 2000, 4000, and 8000 Hz should suffice for most spaces.

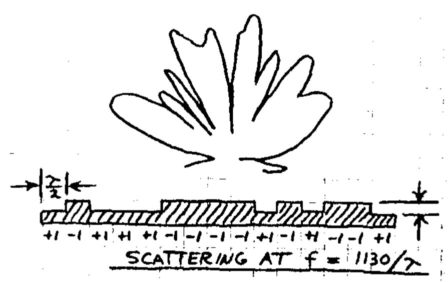


2d. Polygonal

At lower frequencies other more conventional room geometrics take over and at higher frequencies surface roughness itself begins to play a key part.

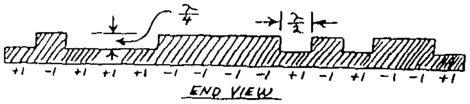
We're particularly eager to hear from any graduates using their TDS equipment in this type of research.

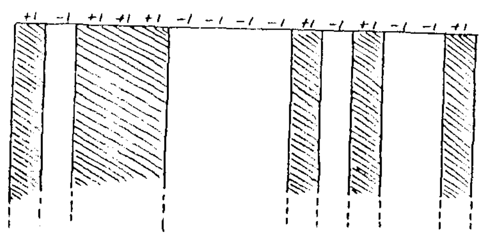




3a & b. Effect of blocking one groove in panel

PSEUDO-RANDOM MAXIMUM-LENGTH CODE N=15, AS USED BY SCHROEDER FOR DIFFUSION





PLAN VIEW

$\frac{\lambda}{4}$ = ONE QUARTER WAVELENGTH OF FREQ f

$\frac{\lambda}{2}$ = ONE HALF " " " " "

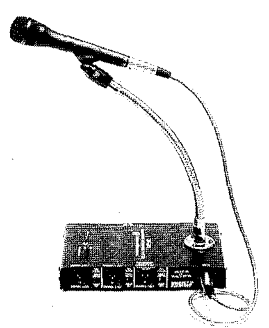
Useful Products

Our grads tell us of products that they find helpful and suggest that others might like to know about them too. Such products are listed below.

We receive an amazing number of product announcements in our mail. Many have a timeliness about them that makes us reluctant to hold them until we have time for a detailed write up. Listed here is a group of products that have caught our attention and are worthy of your investigation:

Crown CM-230 Tridundant Condenser Microphone

The Tridundant condenser is three microphone capsules in a single package. It is a clever and well engineered system inspired by John Royer, head sound man for the Indianapolis 500 motor race track, to mention only one of the many hats he competently wears. Crown International, PO Box 1000, Elkhart, IN 46517. Ph. 219-294-8300.



Audio & Video Switching Modules from FSR

FSR has a new series of EV-4 audio and video switching modules. The audio module is either mono or stereo. Units like these can solve many installation problems in expensive home entertainment systems. Also a Dual Programmable Volume Control, VCA-2. These components take away a remarkable amount of custom design work. FSR, 244 Bergen Blvd, West Paterson, NJ 07424, Ph 201-785-4347.

The Russian Dragon New Signal Alignment Device

The Russian Dragon is a clever device that indicates if your synchronization is rushing or dragging in relative delay. It's a useful device which offers an economical solution to measuring how well signal synchronization has been solved. For more information on the Russian Dragon, contact D. B. Weiss at Signet Sound, 115 East 87th St, Suite 10A, New York, NY 10128. Ph 212-348-9335.

Renkus Heinz New Full Range 3-Way "Smart System"™

The Renkus-Heinz model C-2 full range 3-way "smart system" is the first of a new generation of "total systems concept", TSC, components for both portable concert sound use and permanent fixed installations.

The C-2 is a processor controlled, coaxial, constant beamwidth, horn loaded, phase coherent package designed to be used alone or in arrays.

BIAMP- American Sound Craftsmanship

BIAMP has a new line of consoles of which the Columbia series is enjoying the fastest sales growth in BIAMP's history. The new literature from BIAMP on the Columbia, Cascade, Olympia, and Newport will surely win prizes for beauty and sales appeal.

Portable Automatic Mixer from Shure Brothers

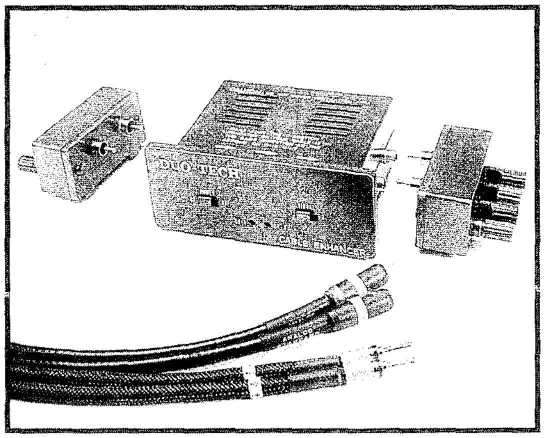
The Shure FP410 and FB 410E is transformer coupled in and out and with Shure's usual integrity in terms of proper level terminology.

Credibility Pending

Do your cables "glare"? Do they "ring"? Do they lack coherence and have ring around the collar? Those of you having cables without a "basic tonal signature" and dynamics can now go into overdrive ecstasy.

The Model CE 1000 Cable Enhancer let's you exercise your cables with "Digitalized rocket sound with rich harmonics" and "relax its stresses" while "conditioning the different skin depths."

Jeff White of White Acoustic Labs in Jeffersonville, IN told us that they saw the product at the CES show.



MODEL CE1000 CABLE ENHANCER™
For <u>Fast</u> Cable Burn-In! (Patents pending)

When asked, "What does it actually do?" they responded, "We are not exactly certain, but it sounds better after you do it." Not only are "patents pending" but so is "credibility".

Dorian Recordings Reaches the Big Time

Dorian Recordings of Troy, NY has entered the arena of major recording companies of classical music. By major we mean of first quality and significant orchestras.

Dorian has just released a CD recording of Prokofiev's Scythian Suite and Stravinsky's Le Sacre Du Primtemps played by the Dallas Symphony Orchestra under the baton of Eduardo Mata.

The recording is made in one of the best of the modern halls designed by Russell Johnson -- the Eugene McDermott Hall at the Morten H. Meyerson Symphony Center in Dallas.

McDermott Hall is a 2062 seat hall with superb acoustics (utilizing the concept of a faster initial decay followed by a second longer slope as the level decreases and the time increases.) We have reported on this hall in an earlier Newsletter (17N2) as an exceptionally good hall.

Dorian's recordings of the Dallas Symphony in this setting is also exceptional and demonstrates the fulfillment of all the promise inherent in their other recordings of smaller ensembles.

In our opinion, Dorian has presented us with a technical tour de force with the fidelity of their recording and their remarkable ability to capture the acoustic flavor of the hall. We will leave artistic judgment to those of you equipped to make that judgment but we found the conductor's reading of the two pieces musically thrilling.

If you have followed the growth of this remarkable recording company you will be totally delighted with this recording and be looking forward with us to a continuing flow of musical quality of the highest order.

Professional Services

Acoustical Consultants may list their cards on this page. There is no charge. The only requirements are that you are a full-time consultant, that you have attended a Syn-Aud-Con seminar, and have an active subscription to the Syn-Aud-Con Newsletter. If you would like to be on our Consultants page, send in four (4) business cards for our file.

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Principal

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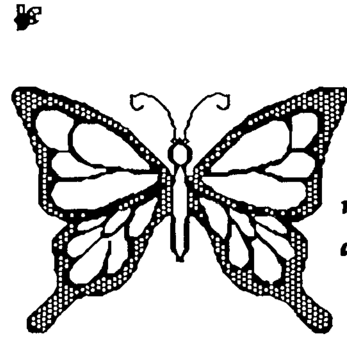
David L. Klepper
Consultant in Acoustics

KLEPPER MARSHALL KING ASSOCIATES, LTD.

FOR SALE: Loftech TS-1 combination audio generator, frequency counter and dB meter, unused since factory update and calbration. Mint condition $215 ($360 new); Fluke 8020B DVM, excellent condition $175 ($304 new). CONTACT: Don Creevy, 4370 Alpine Rd., Portola Valle, CA 94028. (415) 851-0140 evenings and weekends.

POSITION WANTED: Seeking a position to usefully utilize my many years of training and experience. Training: Candidate of Technical, (American equivalent of PhD in Radio/Electronic Engineering). Experience includes 27 years as a test & research engineer, Moscow audio (USSR); Development of calibration tapes, test systems & new methods of audio measurement; Testing of magnetic tapes, tape recorders and various audio equipment; Research of distortion and noise. Contact: Boris Kollender, 2101 Unruh Ave., 1st Floor, Philadelphia, P?A 19149, Ph: 215-333-0978.



If I cannot be free, to do such work as pleases me, near woodland pools and under trees, you'll get no work at all; for I would rather live this life and die a begger or a thief, than be a working slave with no days free.

William H. Davies
(1871-1940)

July Farm Seminar 1991

SYN-AUD-CON SPONSORS

Syn-Aud-Con receives tangible support from the audio industry. Nineteen manufacturing firms presently help underwrite the expense of providing sound engineering seminars. Such support makes it possible to provide the very latest in audio technology while maintaining reasonable prices relative to today's economy and to provide all the materials and continuing support to all 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 grad needs.

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

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