SYNERGETIC AUDIO CONCEPTS

SYN AUD CON

# newsletter

P.O. Box 669, San Juan Capistrano, CA 92693
Ph: 714-496-9599

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SUMMER, 1981

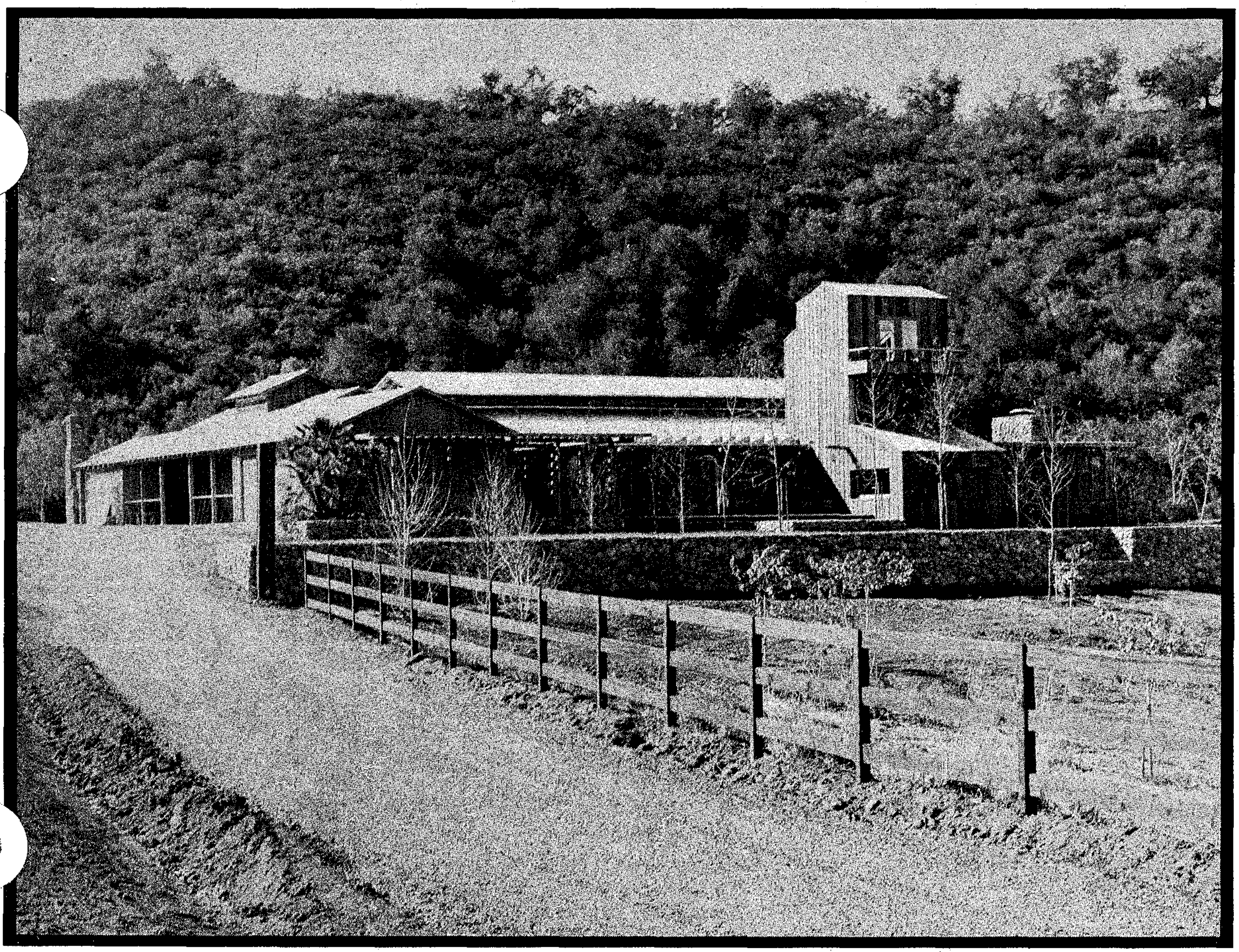


SYNERGETIC
Working together; co-operating, co-operative

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 |



## SYN-AUD-CON AUDIO INDUSTRY SEMINAR CENTER

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# SYN-AUD-CON IS TRAVELING

Syn-Aud-Con is back on the road conducting traveling seminars after an absence of nearly two years. During those two years we built and equipped a unique West Coast Seminar Center high in the Santa Ana Mountains of Southern California on a ranch located within a 40,000 acre wilderness area. Dramatic advances in TEF™ technology, PZM™ techniques, LEDE™ design concepts, SBA distribution systems, and many, many other audio subjects have been incorporated into Syn-Aud-Con's classroom demonstrations.

Yes! We have been listening. Those attending the special classes at our new West Coast Seminar Center have repeatedly told us that the ideal situation would be for us to hold our special workshops at the ranch and still conduct "on the road" introductory classes for those not yet ready for special workshops.

Starting this Fall (September) Syn-Aud-Con has scheduled classes designed to update those who have not had the opportunity to travel to our West Coast Seminar Center. The seminar center will be operated during the Winter and Spring for special workshops on TEF™, financial planning, microphone applications, LEDE™ control rooms, etc.

The "on the road" classes will provide all new update information on TEF™ measurements and their practical application to professional sound situations such as the adjustment of time delays, location of echos, and alignment of arrays. The latest PZM™ technology and applications will be demonstrated using the TEF™ equipment. SBA distributed systems will be shown and demonstrated. Important new loudspeaker array design techniques will be shown utilizing creative use of digital time delay devices.

We have found that sound doesn't travel in a straight line as has been commonly assumed for decades. The energy time curve (ETC) measurements (within the time energy frequency (TEF™) system) clearly revealed that sound from a distant array in a major auditorium was taking a longer path than expected. The air conditioning equipment had generated temperature differentials that refracted the acoustic signal into a curved path. Naturally, the digital time delays required adjustment accordingly (about 5 msecs, which is enough to be crucial in most installations).

If you have not attended a Syn-Aud-Con class in the past two years, these "on the road" classes will be a revelation to you in terms of the acceleration of design techniques in the practical application of audio equipment. The almost unbelievable proliferation of new understanding of what constitutes a good or bad acoustic that is flowing from the use of TEF™ measurements will quickly convince you that a tool of far greater fundamental importance than the 1/3 octave real time analyzer is now at hand.

We'll look forward to sharing these marvelous new tools, insights and techniques with you during one of the Fall 1981 classes.

As a Syn-Aud-Con graduate you will be able to attend a Sound Engineering Seminar at a city near you or at our West Coast Seminar Center or one of our Workshops. The happiest aspect of our new expanded classes is that we hope to see many of our graduates during the coming year.

*Syn-Aud-Con graduates are capitalized throughout Newsletter.

# SYN-AUD-CON 1981 SCHEDULE

## NATIONWIDE SOUND ENGINEERING SEMINARS

| | | | | | |
|---|---|---|---|---|---|
| Denver Area ☐ | Sept. 1-3 | Cleveland Area ☐ | Oct. 5-7 | Atlanta Area ☐ | Nov. 9-11 |
| St. Louis Area ☐ | Sept. 16-18 | Washington, DC Area ☐ | Oct. 20-22 | Orlando Area ☐ | Nov. 18-20 |
| Chicago Area ☐ | Sept. 28-30 | New York City Area ☐ | Oct. 27-29 | Dallas Area ☐ | Dec. 1-3 |

### 1982 WEST COAST SEMINARS
San Juan Capistrano, CA

Feb. 2-4 ☐ Sound Engineering Seminar

Mar. 23-25 ☐ Sound Engineering Seminar

April 20-22 ☐ Sound Engineering Seminar

### 1982 WEST COAST WORKSHOPS
San Juan Capistrano, CA

Designing Loudspeaker Arrays

Designing LEDE™ Control Rooms

Instrumentation

Microphone Application Techniques

Financial Management

# SYN-AUD-CON WORKSHOPS AT THE WEST COAST SEMINAR CENTER

The most beautiful time of the year to visit California is during the Winter months. While it is possible to encounter heavy rains during the winter season, the reward after a rainstorm is a 150 mile view of the entire Los Angeles basin ringed by majestic mountains and with large islands clearly in view out in the Pacific Ocean. Especially in the high ranch country where our seminar center is located the streams are flowing, the grass is green and the vistas endless.

Our first of the newly planned special workshops will be held in January, 1982. Five of these special workshops are planned at the present time. They are:

1. Designing Loudspeaker Arrays: This workshop will be headed up by one of the finest minds we have encountered in this work - DR. EUGENE PATRONIS. Three full days of real practice at how to do it, including working directly with key components.

2. Designing LEDE™ Control Rooms: We have an unusual and diverse team selected to teach these three days. You'll leave really understanding how to do it.

3. Instrumentation: Details to be announced later this Fall.

4. Microphone Application Techniques: We will have internationally recognized recording engineers helping in this class. A real opportunity to hear A-B comparisons between rival techniques as they are demonstrated using a "live" group.

5. Financial Management: This workshop will cover the financial aspects of small, medium and large audio contracting firms. It will be staffed and organized so that small companies work on their specific problems in one meeting while large firms find answers to their problems in a parallel meeting. A workshop developed so you can find specific answers to specific problems and not just hear another talk related to some area of contracting that you are not in.

These workshops will vary in cost, depending on the number of instructors for the workshop, but we would expect that our past workshop price will be the average - $750/person including meals and hotel room.

# MAY 1981 AES REPORT

## Design Aspects of Graphic Equalizers

We have often remarked in class that the most skillfully trained "circuit" engineer is the one most likely to end up out in "far left field" when asked to evaluate the best circuit parameters for use in the design of an equalizer to be used in the adjustment of electroacoustic transducers to their acoustic environment.

This was proven true once again in a paper entitled "Design Aspects of Graphic Equalizers" written by two engineers very conversant with electronic parameters but absolutely naive about acoustic parameters.

The paper incorrectly states that bridged-tee filters are not necessarily minimum phase rather than the obvious that bridged-tee filters can be misdesigned to not be minimum phase. This error was repeated verbally in strong language during the presentation of the paper by one of its authors. R. R. Cordell, of the Bell Telephone Laboratories, corrected the misstatement after the paper by pointing out that correctly designed (conjugate legs) bridged-tee filters, known as "Bode" filters, are indeed minimum phase.

Mr. Cordell's comments as he best recalls them were:

> *That the conventional bridged-tee filters widely used in audio and telephone applications always have the series and shunt elements chosen as electrical conjugates of one another, i.e., parallel arrangements in one will be series arrangements in the other, capacitors in one will be inductors in the other, and resistances in one will be conductances in the other. Such equalizers are a form of so-called Bode-type variable equalizers and generally exhibit well-behaved minimum phase transfer functions.*

The paper's bibliography reflects only the most casual research into the problem at hand. The presenter's cutting remarks about the obscurity surrounding the subject might have been modified had he examined the original sources such as Terman, Bode or Morse.

Band-pass filters are discussed for equalization throughout the paper in spite of their total unacceptability to users who understand the true nature of the acoustic signal.

We remain hopeful that the evident capability for detailed analysis that these authors seem to possess will be exercised more fruitfully in a rewrite of this paper.

## Mel Sprinkle Fellowship Award

There are authorities and then there *are* authorities. To Syn-Aud-Con the title "authority" is all too often misused. Many using the title are self-appointed or part of a clique of self-reinforcing awarders of titles to each other.

Normally, peer group technical societies operate on purely political grounds and only rarely recognize genuine talent or authority. Once a decade a "Dick Heyser" is recognized (but not necessarily for the right thing) and this year the AES Convention in Los Angeles transcended politics--as usual--and gave a fellowship in the AES to MEL SPRINKLE. The official writeup in the AES banquet program reveals the tip of the iceberg in Mel's case.

> *MELVIN C. SPRINKLE, a native of West Virginia, received his Bachelor's degree with honors from Shepherd College in Shepherdstown, West Virginia, majoring in physics and mathematics. He did post-graduate work in radio-electronic engineering at the RCA Institutes, N.Y.; in advanced mathematics and engineering at George Washington University and in acoustics and noise control at the University of Missouri and at M.I.T. under Dr. Leo Beranek. Subsequent to work with the Bureau of Ships, Navy Dept., during World War II, he was employed by Altec Lansing Corp., where he assisted the late H.S.Morris in promoting high quality music in the home using professional audio components. With the exception of several years with Page Communications Engineers, he has been associated with professional audio, including employment with the audio division of Ampex and work as an audio-acoustical consultant and teacher. Mr. Sprinkle is the author of magazine articles and professional papers on audio and acoustical subjects, some of which have been presented at AES' conventions. He has contributed to the professional and mathematically disciplined design of sound reinforcement systems, often collaborating with Don Davis. One of the sound system designs is installed in the N.Y. Giants' football stadium in N.J. The full-range loudspeaker cluster at one end radiates 12 kilowatts of audio power. The system has 12 signal delay zones and, with supplemental loudspeakers, radiates 30 kilowatts. The design methods was the subject of a paper delivered before a New York AES convention. At present, Mr. Sprinkle is a member of the faculty of the Eastman School of Music, Rochester, N.Y., teaching in recording seminars. He is also a faculty member of Capitol Institute of Technology, teaching audio and electronic engineering courses. He is engaged in the design, installation, supervision, and testing of public address systems used in the Washington, DC METRO subway. He is a Registered Professional Engineer in the District of Columbia, Maryland and New Jersey. As a member of the original Sapphire Group, he participated in the formation of the Audio Engineering Society and became a charter member. He has served the Society as chairman of the Washington, DC section and as an active member of the AES Standards Committee.*

Syn-Aud-Con, along with a myriad of others who respect Mel's accomplishments, remarked upon hearing of the award, "It's about time." A group of Syn-Aud-Con graduates joined us at the banquet hall in time to applaud this well-deserved, long overdue recognition of a real audio pioneer who's still very much in the forefront in audio today.

Continued next page....

## Confusion In Academia Re PZM™

Two relatively unknown professors from Canada gave a paper on "The Acoustical Behavior of Pressure-Responding Microphones Positioned on Rigid Boundaries - A Review and Critique" at the recent Los Angeles AES Convention.

The rigidity of their thought on the subject and its consequent biasing of their investigation resulted in a carefully researched investigation into all of the already published facts about pressure zones and devoid of any new insights.

The PZM™ technique *is* a new insight into the use of such pressure zones but these investigators failed completely to uncover it. Some of the reasons for their failure are:

1. They assumed that the designers of the PZM™ are fools who didn't understand the fundamentals of acoustics.

2. They built and tested a non-PZM™ microphone and considered it a PZM™.

3. The tests were performed with a major error regarding the source used (an error demonstrated in every Syn-Aud-Con class, yet they failed to detect it).

4. At one point in their tests of flush mounted microphones the problem that PZM™ uniquely solves appeared in their data but they missed it completely.

Ed Long's comment on their performance perhaps best describes their shortcoming--"Their failure to recognize the basic accomplishment of the PZM™ technique should strengthen the patent claims."

As a Texas friend recently told us, "There are few more fearsome sights than ignorance in action."

## Sound Reinforcement Session

As Syn-Aud-Con graduates quickly learn, the parts of an AES Convention that are always of interest to them are those technical papers given by other Syn-Aud-Con graduates. Further, there are those unscheduled technical papers given extemporaneously up in the Syn-Aud-Con suite when extremely talented guests and graduates exchange views on as yet unsolved audio and acoustic problems.

Syn-Aud-Con graduates have provided some genuinely innovative papers on the "re-mapping" of linear space to angular space. Loudspeaker coverage in terms of required elevations, azimuths and ranges can be accounted for via programmable calculators which generate special plots rather than having to build physical models and using light projectors, etc. TED UZZLE and FARREL BECKER both made valuable contributions to these techniques (first suggested by Ed Seeley at the May 1978 AES Convention. TOM McCARTHY of Northstar in Minneapolis further developed the concept in his AES paper in New York in 1978).

DR. EUGENE PATRONIS and his student at Georgia Tech, Catharina Donders, placed in the audio literature a "housekeeping" technique for keeping track of the allocation of total acoustic power in terms of direct-to-reverberant ratios. We believe this elegant solution will be adapted to many users' needs and become the basis for a vastly improved approach to array design.

TED UZZLE (now of Altec) presented an extremely interesting loudspeaker coverage paper and FARREL BECKER gave a practical paper on coverage from presently available data. It became evident that the audio world is about to solve, and solve elegantly, that most difficult of systems problems--the design of complex loudspeaker arrays. We have waited a long time for the advent of experienced practical talent thoroughly based on *usable* theoretical parameters to come to grips with how to engineer large complex arrays.

Dr. Patronis explaining a concept to Farrel Becker and Mike Hoover during the February Heyser class.

DAVE KLEPPER, Session Chairman, as always, contributed heavily to the general discussions on loudspeaker array design from a wealth of actual jobs in service. Dave's experience bank account is one of the largest in our industry and, because of his receptivity to new ideas, he has a record of steady improvement throughout his career.

When the Patronis-Donders technique is wedded to the Uzzle-Becker calculator programs, there emerges the elimination of physical model making while accurately plotting and understanding the loudspeaker coverage patterns required and their admissibility in terms of articulation losses incurred.

Continued next page.....

MAY AES REPORT continued.....

Syn-Aud-Con is very excited about these progressive steps. We anticipate a remarkable improvement in sound systems as soon as this information is made available and is correctly understood by those engaged in the design of today's large complex loudspeaker arrays. That there is a great deal of educational effort still required was made evident when an AES workshop leader (as reported back to us by those attending his workshop) didn't know the difference between loudspeaker directivity factor Q and its coverage angle $C_{\angle}$. But then, as we said at the beginning of this article, it's the Syn-Aud-Con graduates who have accepted the challenge of solving these problems.

RUSS BERGER of Highgrove House in Dallas gave an excellent tutorial paper on the design and application of Helmholtz resonators in control rooms and studios.

## Comments on Spring AES Meeting

This May's AES was, in the judgment of a surprisingly large number of attendees, one of the poorest managed conventions in recent memory. The papers chairman allowed some authors as many as three papers while eliminating worthwhile papers from authors not known to him. The loudspeaker system used in the technical sessions looked like a direct copy of the 1928 RCA photophone theater system with sound in keeping with artifacts of that era.

It has become increasingly apparent in recent years that the AES is operated primarily for the revenue obtained from manufacturer's exhibits at the convention. A clique of behind-the-scenes collaborators is gradually building an internal bureaucracy that is increasingly less responsive to either the membership's needs or desires.

The Editor of the Journal is without peer in his total lack of understanding of audio, interest in audio, or contributions to audio.

This year's technical sessions witnessed new lows in papers that richly deserved burial elsewhere and a number were deliberate attempts at obscuration of the facts. (Ignorance is expected and accepted because we all need a forum wherein we can grow--but deliberate misleading is, or should be, a NO-NO!)

The fact is that the society hierarchy now has literally over a million dollars to play with and they do not have the good sense to listen to the exhibitors or the representative membership as to desired directions the society should seek. The Syn-Aud-Con Newsletter would be interested in knowing how many of its readers feel that it might be time to start a new society dedicated to those in the professional audio field--a society that would develop qualification tests and issue professional certification of its members who take and pass the test. Perhaps it could be called the Society of Professional Audio Engineers (SPAE).

Drop us a line if you agree (or disagree) with the above analysis. Syn-Aud-Con *is* responsive to its supporters and we are very interested in your viewpoints on this subject. Our own position on this matter is not solidified and your response will indeed help us evaluate what if any action Syn-Aud-Con should support in the future regarding the AES.

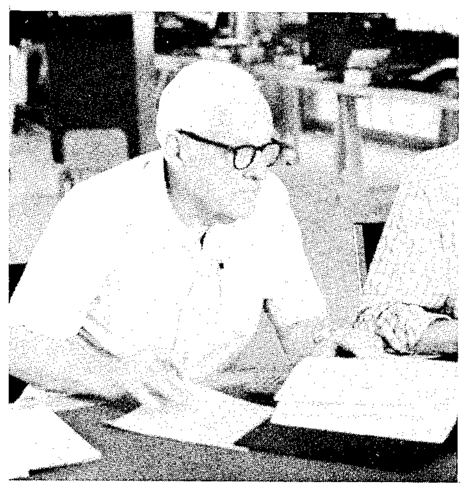
## SYN-AUD-CON HERO

During World War II Lieutenant Commander STEVEN H. SIMPSON, JR., played a key role in the OSS's penetration first into Europe then into Germany itself. He developed a unique radio system for his agents and flew in the early missions that tested it.

His colleagues variously found Steve "cheerful," "always smiling," "fiercely independent," "a very collected human being," "a belligerent man masquerading as a Naval officer."

Steve is still going strong, has mellowed a little, and is still one of the most interesting and rewarding human beings it has been our privilege to know. Steve is the subject of an entire chapter in the book *PIERCING THE REICH* written by Joseph E. Persico. This book, written with the cooperation of the intelligence agencies involved, tells of how "some men would die on invasion beaches, and others against a wall." It's a book about dedicated, tough-minded, patriotic men old enough to fully appreciate the risks, better informed than many in the military of the penalties of failure in their specific cases, and brave enough to go ahead anyway.

This photograph of Steve Simpson was taken at our June 1981 Syn-Aud-Con class at our West Coast Seminar Center. Heaven help the design problem Steve has his hands on. He just looks them straight in the eye and proceeds to eliminate whatever is in his way of a successful conclusion.



Steve Simpson - 4-time Syn-Aud-Con graduate.

# MAY 7-9, 1981, CLASS

# LEDE™ DESIGNERS

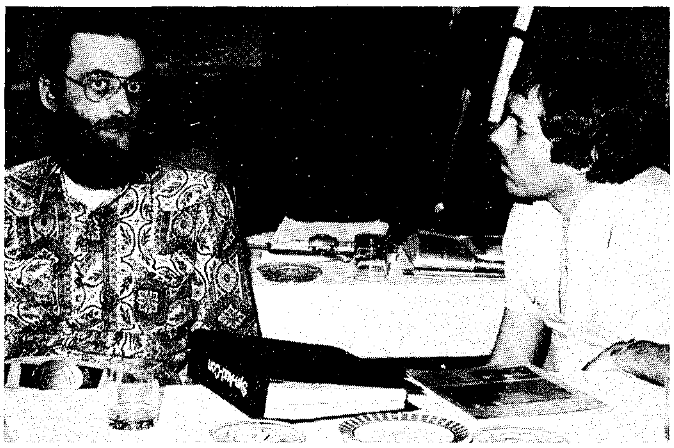


FIG. 1: Ed Bannon and Chips Davis.

FIG. 3: Glenn Meeks during the measuring session at Chips Davis' studio with the first Heyser TDS class in 1979.

There are numerous LEDE™ type control rooms being built all over the world. Here in the United States there are three very active control room designers actually constructing full LEDE™ rooms and helping to materially advance both our knowledge of LEDE™ design and how to use TEF™ measurements in the most effective manner during the construction and "proofing" of these rooms.

The premier team is CHIPS DAVIS and ED BANNON in Las Vegas, Nevada. They are currently involved in helping studio owners in New York, Caracas, Oakland, and San Rafael, California.

FIG. 2: Russ Berger (center) showing members of the February 1981 TEF™ class one of his HP41C programs.

RUSS BERGER in Dallas is quite busy in proving his capabilities (which are considerable) in the Texas area.

GLENN MEEKS of Indianapolis has developed a unique approach that includes retro fitting older rooms via TEF™ analysis to achieve striking aural improvements undreamed of prior to our ability to see what we are hearing.

Letters from overseas indicate activity that unfortunately does not have the benefit of measurement. While we all know that even a poorly done LEDE™ control room significantly out-performs even the best conventional rooms, it's a shame if entire areas abroad miss out on the really superlative advancements that Chips, Russ and Glenn can provide with their instrumentation and increasing experience.

FIG. 4a: Hellmuth Kolbe

An exception is HELLMUTH KOLBE of Switzerland who attended the February 1981 Heyser class. He went back to Switzerland and ordered a complete B & K TDS-ETC laboratory--well over $50,000 U. S. dollars. That's true commitment! He said he couldn't afford to wait for the new Crown instrumentation.

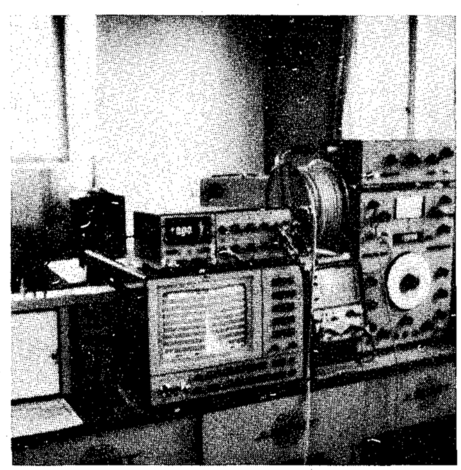


FIG. 4b: Kolbe's new B & K laboratory.

# LANDMARK AES PAPER by C. A. "PUDDIE" RODGERS

C. A. "PUDDIE" RODGERS' remarkable paper, "Pinna Transformations and Sound Reproduction," received publication in the April 1981 *JOURNAL of the AES*, Volume 29, No. 4, pages 226-234.

This paper contains that extremely rare combination of a daring intuitive approach, unbelievably thorough academic scholarship, and important new information on a previously misexplained phenomenon.

Like all truly great papers, Puddie's is solidly founded on the early literature in the field. Her bibliography is a necessary preliminary education, not an exercise in name dropping.

Her approach is tutorial, as it must be, because they didn't get it right the first time.

Of absolutely fundamental importance to all of us is the identification of the role of comb filters in the assignment of azimuth and elevation to sounds the human listener receives.

The abstract from the paper quoted below gives the clue to the paper's importance. Note Puddie's use of TDS. Puddie attended the second Heyser TDS-ETC Workshop in November 1979. When she saw the comb filters on the TDS generated by misaligned loudspeakers, she remarked that it gave her an idea for future research. Her paper at the May 1980 AES Convention was the result of this research.

Puddie Rodgers at the November 1979 Heyser TDS Workshop talking with Chips Davis.

*Many studies have shown that the pinnae transform incoming signals, superimposing upon the original signal a comb-filter-like spectrum. This spectral shaping has been shown to add an additional cue to the now classic hierarchy of localization cues: interaural intensity, phase, and time of arrival differences. Recent evaluations of misaligned*[1] *loudspeakers using time delay spectrometry reveal spectral shapes which are strikingly similar to pinna transformations. The implication is that misaligned loudspeakers, poorly placed microphones, or other early reflections introduce spectral aberrations which may be decoded by the auditory system as cues to source position. The possible consequences of the pinna transformations to the interpretation of psychoacoustic phenomena such as auditory imaging, the cocktail party effect, and the precedence effect are discussed.*

[1]*The term misaligned as used in this paper refers to drivers whose acoustic centers are at different distances from the listener.*

We believe that only a very few years will be required before Puddie is recognized as having produced a real breakthrough in our understanding of the human hearing process, especially as related to the judgment of electroacoustic devices. We are particularly grateful that all attempts to "appropriate" her work or to "despoil" it by peer review has been held to a minimum.

We salute the talent that went into this research and expectantly await its full development as experience is added to her very visible intellectual achievements. Knowing Puddie's single-minded devotion to audio and acoustics, we're sure whatever project she tackles next will result in important new knowledge available to us all.

One last point. Failure to read and study this paper will severely handicap any sincere inquirer into the behavior of loudspeakers, as well as those interested in truly understanding why small "dead" rooms sound like they do. Contained in this paper is the "Rosetta Stone" to achieving practical "surround sound out of two channels" with a realizable device. What more can we say?

# A FEW CAMEO DEFINITIONS

1. Ohms law $P = \frac{E^2}{Z}$
2. EIN - the lowest EIN is -124.8 dBm
3. White noise energy increases 6 dB/oct

While this dictionary is the output of a sincere effort to clarify audio terms, it succeeds in just the opposite result. The definitions are either meaningless, distorted, or plain incorrect. The biggest mystery is how these definitions got by the engineering staffs of the Cameo companies listed in the rear of the book.

# NEW TEF™ LICENSEES

TEF™ licensing continues. Once the dedicated TEF™ analyzer is released to the market, there will be no further TEF™ licenses issued as the license fee then becomes a royalty built into the analyzer's base price. The advantage of a separate TEF™ license is that it allows the holder to build and use one TEF™ analyzer (including legal add-ons and modifications to a dedicated one if he wishes).

The user of the dedicated unit is legally restricted to that unit alone, though at the present time that doesn't seem that it will be much of a limitation.

We now estimate that our issuance of TEF™ licenses under this program will cease about January, 1982. (We stop whenever Crown sells their first analyzer to the public.)

We also continue to feel that possessors of original TEF™ licenses have historical proof that they were there when it happened. Here are two new licensees:

| | |
|---|---|
| Mr. Brian G. Wachner<br>BGW Systems, Inc.<br>13130 S. Yukon Avenue<br>Hawthorne, CA 90250 | University of California<br>Lawrence Livermore Laboratory<br>7000 East Avenue<br>Livermore, CA 94550 |

# PROBLEMS IN THE MEASUREMENT OF REVERBERATION TIME

Mr. Robert Hagenbach, a recent TEF™ licensee in Indianapolis, Indiana, sent us a copy of a really outstanding paper on "Problems in the Measurement of Reverberation Time" written by Theodore J. Schultz (BB&N) back in the October 1963 Journal of the AES, Volume II, No. 4, pages 307-317.

It is easily the most sensible paper it has been our privilege to read on the subject of reverberation time measurement.

We have remarked many times that what is read on a reverberation meter in a small "dead" room is not the decay time of the reverberant sound field and asked the rhetorical question, "Just what is the meter measuring?" This paper supplies the answer.

> *In the case of a small room the microphone, instead of responding to a random sound field (as required for the validity of the theory on which these methods depend) will delineate a transfer function of the room......This is a curve which gives a great deal of information about the structure of the room response in terms of modal frequencies, but only a little about the absorption on the boundaries.......It does not provide a valid measurement of the reverberation time in the room, however.*

The paper includes vastly more than this simple excerpt indicates, all of it of genuine interest to anyone who is serious about understanding the difference between rooms where classical statistical analysis is useful and rooms where it is not.

Mr. Schultz was far in advance of his contemporaries. Current researchers using the ETC measurement will find that he supplies some excellent theoretical underpinning to the data they are observing. Our special thanks to Mr. Hagenbach for finding a paper of this magnitude that we must confess we completely missed when originally published.

# NOT AMPLIFIER GAIN, AMPLIFIER SENSITIVITY

At long last Syn-Aud-Con has run across a suitable name for what the circuit engineers like to miscall gain. They are, in reality, measuring the amplifier's *"sensitivity"* (i.e., How many volts in for how many volts or watts out?) We use *sensitivity* figures for microphones and for loudspeakers so why not for electronics as well If kept to voltages and not turned into decibels (the dB is reserved for gain), then sensitivity ratings can make system setups easy for anyone with only a voltmeter, provided the loudspeaker's true resistance as a load is known so that the voltages can be transformed into dBm at the electrical output of the system.

In using voltage *sensitivities* it is important to remember that the meter lag factor (10 dB) becomes roughly 1/3 the maximum voltage.

$$20 \log \frac{1}{X} = 10 \text{ dB} \qquad \frac{1}{X} = 10^{\left(\frac{10}{20}\right)} \qquad X = \frac{1}{10^{\left(\frac{10}{20}\right)}} = .316....$$

The reason that you don't convert sensitivity figures (i.e., $X$ volts in produces $y$ volts out) into decibels is because the figure that results *does not describe what happens at the output of the system.* It is merely the voltage amplification of the device.

We encountered this concept in *The Tektronix Cookbook of Standard Audio Tests* by Clifford Schrock. This cookbook does not describe gain measurements but does have many very practical hints on how to conduct electronic tests of audio components with proper regard to safety, accuracy, and repeatability.

# $2^{(256)} = 1.1579208 \times 10^{77}$

In Volume II of his *Magia Universalis Naturae et Artis*, Herbepoli 1658, 4to, the Jesuit Gaspare Schotto, having discovered, on some grounds of theological magic, that the degrees of grace of the Virgin Mary were in number the 256th power of 2, calculated that number. Whether or not his number correctly represented the result he announced, he certainly calculated it rightly.

$$2^{(256)} = 1.1579208 \times 10^{77}$$

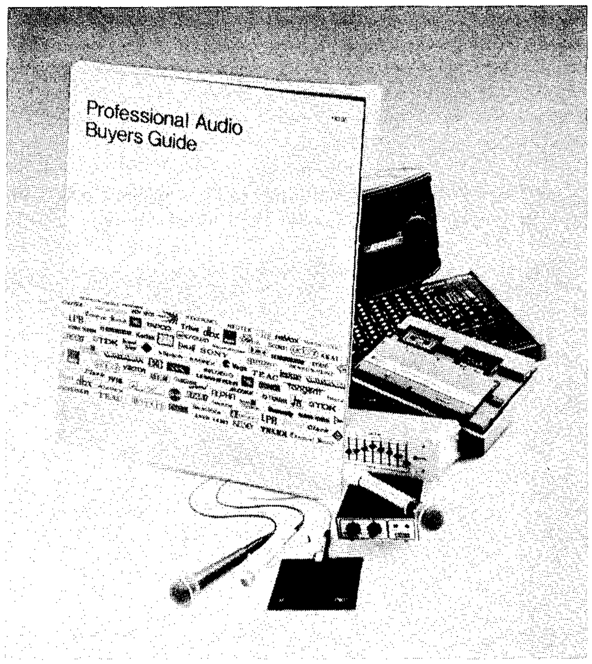
Gaspare Schotto passed on in 1666 which indicates that it was not an overwhelming effort to properly calculate a number with 78 places.

*Assorted Paradoxes*
by Augustus De Morgan

# PROFESSIONAL AUDIO BUYERS GUIDE

HOWARD PARKER of Sound Investment Enterprises in Thousand Oaks, California, has put together a special audio catalog entitled the *Professional Audio Buyers Guide*. Where else could you find the very new products like PZM™ and SBA cataloged but from a many-time Syn-Aud-Con graduate. Made to sell for $15.95, we suggest one of these will be in the hands of the next church committee you meet with.

SIE Publishing has just introduced its new *Professional Audio Buyers Guide*. This is the most complete guide to professional audio sound and recording equipment available. Within its 240 pages are over 70 manufacturers and thousands of professional products.




Now in one single guide you can have at a glance products by Akai, Altec, Anvil, Atlas, Audiotronics, Belden, Berkey, Biamp, Bogen, Beyer, Bose, Community Light and Sound, Crown, DBX, Electro-Voice, HME, JBL, Klipsch, Kodak, Neumann, Otari, Revox, Sennheiser, Shure, Sony, Switchcraft, Tapco, Tascam, TEAC, Telex, Vega, Yamaha and many more.

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Your cost for the *Professional Audio Buyers Guide* is only $15.95. Special quantity, educational and dealer prices are also available. To get your copy of the *Professional Audio Buyers Guide,* see your local sound dealer or contact SIE Publishing, P. O. Box 4139, Thousand Oaks, CA 91359, 213/991-3400.

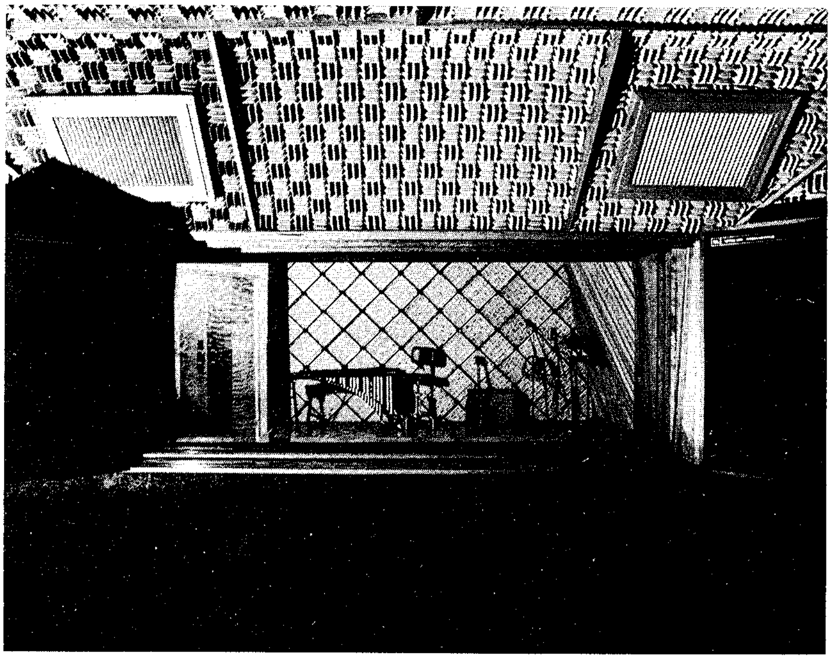
# EMIL BAR FROM ISRAEL

We recently had a studio owner from Tel Aviv, Israel -- EMIL BAR, who attended our June class. We are pleased to know that he will carry back to his homeland the latest ideas in the use of TEF™ measurements and LEDE™ design.

After attending the class at the ranch, Emil flew to Las Vegas for time with CHIPS DAVIS and ED BANNON in their highly reworked LEDE™ control room.

We expect to hear of interesting control room designs in Israel as a result of this visit.

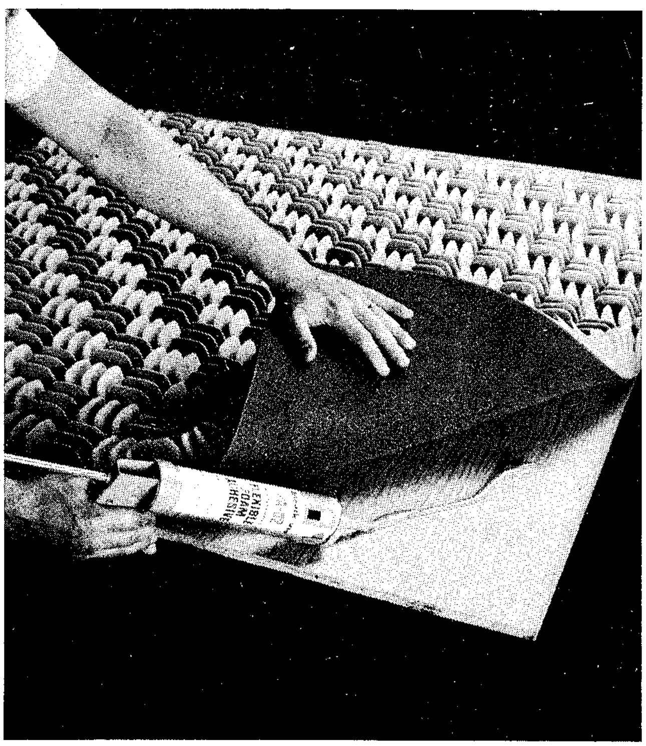
# SONEX FOAM

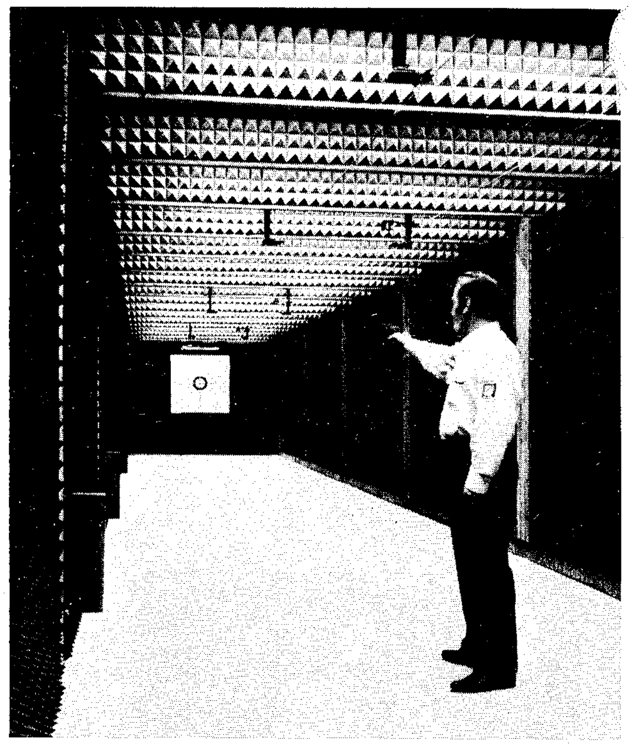


One of the steady inputs we are receiving these days is on Sonex sound absorbing foam. It's finding application in shooting ranges, broadcast and recording studios, as well as control rooms. The photographs illustrate the wide range of successful applications. The ease of application is no small factor and several of these pictures reveal how handy this material is to use.

Syn-Aud-Con has found that 3' X 3' patches of this material are indispensable during TEF™ measurement work as it possesses so much attenuation that it's easy to spot the energy that changes as the Sonex intercepts its path.

One caution with regards to shooting ranges. There is a necessity to provide adequate ventilation in order to avoid lead poisoning from the disintegration of the bullets against the typical back stops. Also, the man in the picture is not wearing hearing protectors and, on an indoor range, *that's bad news*.





We continue to be enthusiastic about Sonex and hope all of you have had an opportunity to sample this most useful material. Alpha Audio is the national distributor (2049 W. Broad Street, Richmond, VA 23220). It was NICK COLLERAN of Alpha who recognized the potential that Sonex has in audio. (The manufacturer of Sonex is in noise control work.) Nick has been the force behind it's use in audio. KEN WAHRENBROCK has made special arrangements with Sonex and Alpha Audio to act as a distributor to Syn-Aud-Con graduates. For further particulars write Wahrenbrock Sound Associates, Ltd., 12115 Woodruff Blvd. #A, Downey, CA 92041 (213) 861-0397.

## MAY 19-21, 1981, CLASS

# AN AUDIO INTRODUCTION TO SCIENTIFIC CALCULATORS USING REVERSE POLISH NOTATION

Reverse Polish Notation (RPN) is my favorite calculator entry technique and I suspect that it will dominate programable audio input devices in the future.

In Syn-Aud-Con classes we frequently use three example problems to illustrate the efficiency and ease of RPN in specific audio calculation.

## EXAMPLE NO. 1

Calculate the shunt resistor to match a 600 ohm filter to a 2800Ω input in the amplifier following the filter. The classic form of the required equation is:

$$R_T = \frac{R_{IN} + R_D}{R_{IN} - R_D}$$

where: $R_T$ is the desired termination shunt resistor
$R_{IN}$ is the amplifier's input resistance
$R_D$ is the desired value

Using a calculator offers much quicker equation possibilities such as:

$$\frac{1}{R_T} = \frac{1}{R_D} - \frac{1}{R_{IN}} \qquad \text{or} \qquad \frac{1}{600} - \frac{1}{2800} = \frac{1}{763.6}$$

Using RPN the problem is solved by: 600, $1/x$, 2800, $1/x$, - , $1/x$ or a total of six keys used.

The old way: 2800, ↑ , 600, x , 2800, ↑ , 600, - , ÷ a total of nine keys are used.

Possession of a calculator will often lead to extensive rewriting of familiar equations that were originally intended for paper and pencil solutions.

## EXAMPLE NO. 2

Generate the "Renard" numbers from 100 Hz to 1000 Hz. "Renard" numbers are equally spaced *logarithmic* intervals that "so-called" 1/3 octave (actually, they are 1/10 decade) equalizers are spaced at.

It is not commonly recognized that the *labels* on these equalizers *are not* the actual frequencies they should be tuned to (though many are).

Using an H.P. 41C, the calculation sequence is as follows: •, 1, SK, $10^x$, ↑ , ↑ , ↑ , (press X 19 times) and 100 appears in the X register.

| Press | Label | Actual Renard Number |
|---|---|---|
| | 100 | 100 |
| X | 125 | 125.893 |
| X | 160 | 158.489 |
| X | 200 | 199.526 |
| X | 250 | 251.189 |
| X | 315 | 316.228 |
| X | 400 | 398.107 |
| X | 500 | 501.187 |
| X | 630 | 630.957 |
| X | 800 | 794.328 |
| X | 1000 | 1000.000 |

If one were to continue pressing the X's key, the second decade from 1000 to 10,000 would be generated.

The advantage of the H.P. RPN system here should be obvious even to the most rabid "algebraic" notation fan.

## EXAMPLE NO. 3

Given an impedance of $16e^{i0.524}$, what is the AC resistance and what is the reactance value?

When one recalls that: $16e^{i0.524} \equiv 16\angle 30^\circ$ because 0.524 is in radians, then by putting the calculator into the radians mode: EXQ, ALPHA, R, A, D, ALPHA

And then using: 0.524, ↑ , 16, SK, P → R you obtain 13.856Ω ACR in register $x$ and $x \rightleftarrows y$ 8.00Ω reactance in register $y$.

Continued on next page.....

Converting back to the degrees mode EXQ, ALPHA, D, E, G, ALPHA

Using $x \rightleftarrows y$ to place the value 13.856 in the $x$ register again, we can then press SK, R → P and we get 16Ω for the impedance Z and $x \rightleftarrows y$ a phase angle of 30°.

CONCLUSION

If you have not yet tasted the freedom of mathematical exploration these remarkable devices offer the self-tutored, our hope is that these basic examples will tempt you to do so. If you are already skilled in their use, we hope these simple exercises may have suggested revisions in older forms for you to examine.

# VERIFYING THE PRESENCE OF A REVERBERANT SOUND FIELD

The measurement of the reverberation time (in secs) for 60 dB of level change ($RT_{60}$) requires that you first have a *reverberant* sound field to measure in.

The definition of a reverberant sound field is:

> "A diffuse or reverberant sound field is one in which the time average of the mean square sound pressure is everywhere the same and the flow of energy in all directions is equally probable. This requires an enclosed space with essentially no acoustic absorption."

Many practical architectural spaces are semi-reverberant but still have much energy from a diffused field present.

A quick practical test for the presence or absence of such a sound field is as follows:

1. Measure the direct sound output level ($L_D$) in dB of the sound source at 4 to 8 feet.
2. Extrapolate by inverse square law out to a distant measuring point.
3. At the distant measuring point, measure the total sound level ($L_T$) in dB.
4. To find the reverberant sound level ($L_R$), subtract $L_D$ at the distant point from $L_T$ in the following manner:

$$10 \log \left(10^{\left(\frac{L_T}{10}\right)} - 10^{\left(\frac{L_D}{10}\right)}\right) = L_R$$

5. See if the reverberant sound level ($L_R$) is above the ambient noise level ($L_{AMB}$) by at least 30 dB (enough to allow a minimum of 20 dB of decay with a S/N of 10 dB at the low level end of the decay).

If the conditions above are met then an $RT_{60}$ measurement can be considered as valid.

# EARlogs

Those Syn-Aud-Con graduates interested in hearing protection programs (and it seems a majority of us are) will be interested in a superior series of technical monographs called "EARlogs" written by Elliott H. Berger and distributed by E.A.R Corporation, the manufacturer of the ear plugs we give out in class.

We found "EARlog" interesting, sensible and thought provoking. You can obtain your set by writing to:

E.A.R. Corporation
7911 Zionsville Road
Indianapolis, IN 46268

# THRESHOLD SHIFT

From Elliott Berger's "EARlog" came the following, to our mind, relevant measurement of threshold shift.

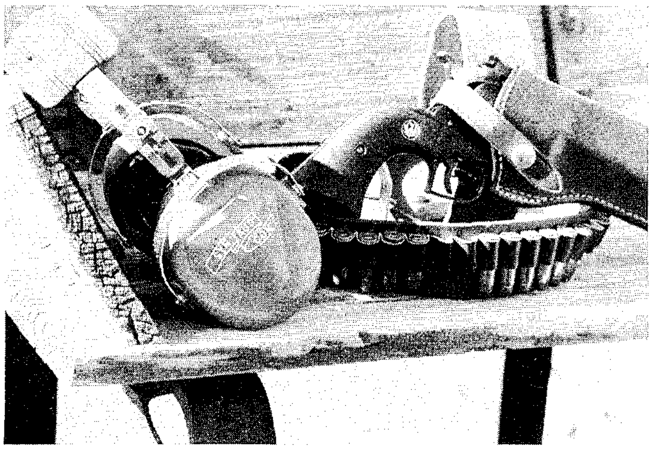
Set your car radio to the lowest detectable audio level just before switching off the ignition and going in to work (leaving the volume control set in place and using the ignition key as the shutoff).

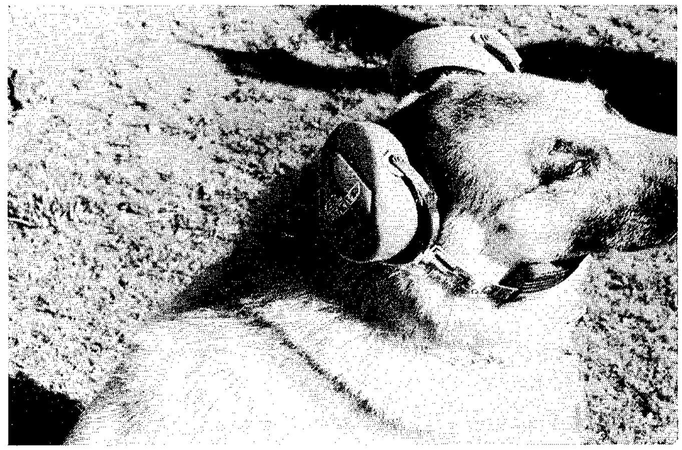
Upon returning to the car after work, turn on the ignition and see if you can still hear the radio. The amount you need to turn it up is a direct measure of the threshold shift (if any) you have experienced during the day.

# SYN-AUD-CON HEARING PROTECTORS FOR SALE

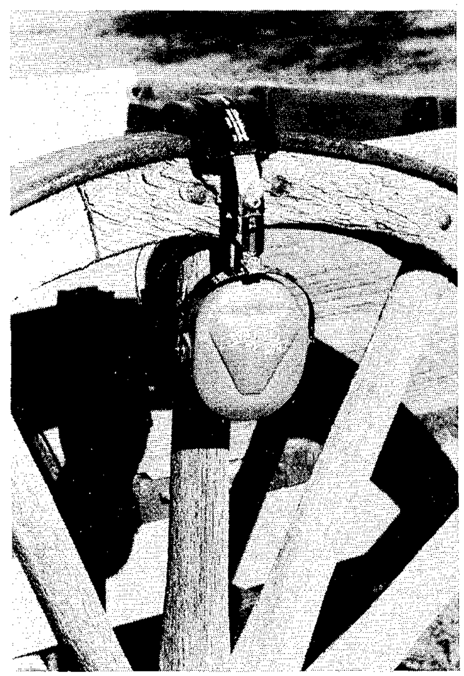
Syn-Aud-Con has again made a special purchase of David Clark Model 27-L "Straightaway" hearing protectors with the Syn-Aud-Con logo on the earpieces. We have sold many hundreds of these hearing protectors in the past and because of the, to our mind, necessity for professional sound men to own a pair, we have reinvested the income from the original sets into a new inventory.

One day we hope that Syn-Aud-Con hearing protectors will be donned in place of displaying Paul Klipsch's "Bull Shit" buttons at AES and elsewhere to express the listener's response to being "put upon" while held as a semi-captive audience. We are reprinting below the writeup that appeared in the Newsletter, Volume 7, No. 2, (Winter 1980).





Quoting from an article in the *AMERICAN RIFLEMAN* of September, 1975, entitled "Gunfire Noise Levels" by William Dresser -- "Well-qualified otologists seem to be in general agreement that approximately 150 dB should be the maximum peak sound pressure limit for gunfire noises without considerable danger of impairment of speech reception. About 140 dB seems maximum for such noises without danger of loss of good hearing of music, etc., and 160 dB about maximum to avoid requirement for payment of compensation for industrial hearing loss."



Some examples are quoted:

1. A 12 gauge, gas operated, 28" barrel shotgun with Cutts compensator:

   172.5 dB PSPL
   Duration: 2.8 milliseconds

2. A 22 caliber short, hollow point, high speed (barrel length unspecified but fired in a rifle):

   157.0 dB PSPL
   Duration: 2.4 milliseconds

3. A .458 Magnum caliber rifle:

   174.7 dB PSPL
   Duration: 2.5 milliseconds

For those Syn-Aud-Con graduates desiring to make such measurements, you require a 1/4" or 1/8" high intensity microphone. Your standard measurement microphones and the time constants in your standard analyzers and sound level meters can't do the job. Recording high speed oscilloscopes are recommended.

The hearing protector offered by Syn-Aud-Con is specially manufactured for us by our sponsor, David Clark Company. It is a model 27-L with a special Syn-Aud-Con imprint on each earpiece.

These protectors are without question the finest we have ever tested, are much lighter in weight and smaller in bulk than our previous personnel protectors and at least 6 dB more effective. Absolutely no disturbance of hearing occurs even when shooting the .458 Magnum. Consequently, we feel confident in recommending these protectors without reservation for audio men desirous of preserving their wide frequency range sensorium.

Our price is $18.40 plus $1.00 for postage.

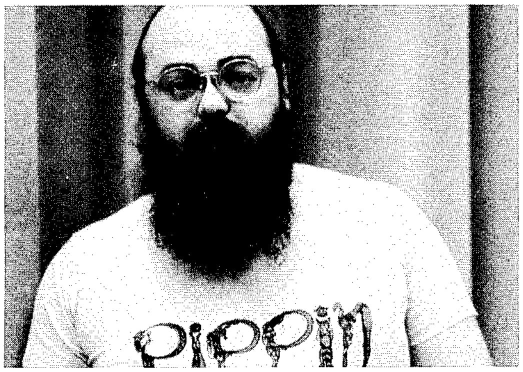
Continued on next page....

SYN-AUD-CON HEARING PROTECTORS FOR SALE continued....

Be sure that anyone you know who enjoys shooting has one of these as part of his shooting equipment. These protectors are so effective in blocking muzzle blast that heavy recoil *seems* markedly reduced. The psychological effects of muzzle blast can be tremendous and the best technique we have ever encountered for curing flinching is to first adequately protect the shooter's ears and then hand him a firearm loaded by another so he doesn't know if it really is loaded or not. The flincher will jerk back a foot or more when he pulls rather than squeezes the trigger. After a few tries with no ammo, quietly slip a live round in and the former flincher will now be hitting the bull's-eye.

With the advent of these superior hearing protectors, Don has been able to indulge his interest in magnum pistols and rifles free of fear of damage to either his nerves or his ears.

## PAN POTS & THEIR USE



JOHN LABERDIE of Dale Ashby & Father, during one of the sessions in the Syn-Aud-Con suite at the Los Angeles AES Convention, pointed out that while you can "pan" a signal back and forth (via amplitude shifts), they will only sound natural when "panned" to the actual geometric location of the original group.

We have no doubt that the comb filter clues PUDDIE RODGERS has identified and which are not materially affected by the amplitude shifts in pan pots continue to inform our mind that the new amplitude location doesn't match the comb filter clue as to the correct location. Result--an unnatural sound to the ensemble being reproduced. John is a close observer of the work of some of New York City's most skilled mixers and we believe he has a point well worth further study on the part of those who build or use pan pots.

## DESIGN EQUATIONS FOR "IN LINE" CEILING ARRAYS

DAVE ANDREWS of Andrews Audio in New York City, glancing at pages 4 and 5 of Newsletter Volume 8, No. 2, (Winter of 1981), observed that by utilizing the first equation on page 4 and the last equation on page 5 you had all you needed in order to successfully design an "in line" array.

1. Find the $Q_{min}$ for the desired $\%AL_{cons}$ and $D_{2ss}$.
2. Examine how you wish to cover the area and if it becomes advisable to use a series of loudspeakers of equal Q and $C_L$ then:
3. Find N by $N = \frac{Q_{min}}{Q_{avail}}$
4. To find the proper $D_{2max}$ (the $D_2$ that allows the selected $\%AL_{cons}$ to be developed with the $Q_{avail}$ chosen)

$$D_{2max} = \left(\frac{D_{2ss}}{N}\right)$$

Dave spotted these simplified relationships right in the middle of a class. That's known as synergy!

## NEW CATALOG

J. W. Davis and Company of Dallas, Texas, has issued a new enlarged catalog for their products. J. W. Davis & Company is the manufacturer of the Heyser Signal Biasing Amplification, SBA, system Syn-Aud-Con recommends to you in our classes. Many of the largest sound contractors in the United States keep this catalog close at hand as one of the most efficient sources of hard-to-find accessories as well as a preferred source of low-cost baffles and loudspeakers. For the beginning contractor in sound systems work their catalog provides a reliable single source for complete modest systems.

All it takes to get your copy is a request on your letterhead. We particularly think you will find their new "Technical Notes" sections of interest for their straightforward relevant data.

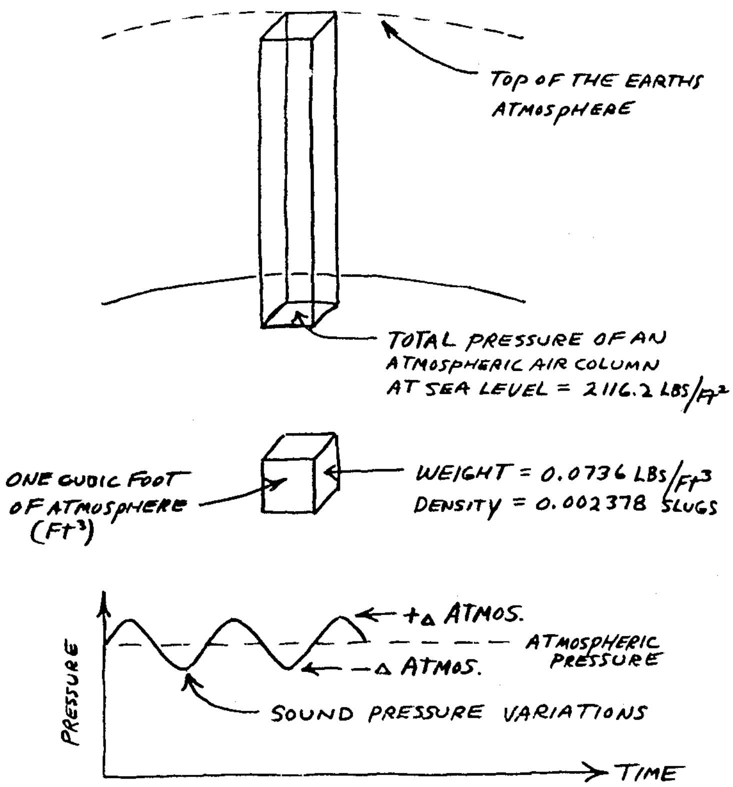
Write: J. W. Davis & Company
3215 Canton Street
P. O. Box 26177
Dallas, TX 75226

Or call: 214/651-7341

# NOTES ON THE ACOUSTIC MEDIA

We live at the bottom of a deep ocean of air. If we were able to isolate a column of air one foot square ($ft^2$) the height of the earth's atmosphere, we would find that the total pressure at the bottom of the column (sea level) would be 2116.2 $lbs/ft^2$ (196.6 $KG/M^2$).

PARAMETERS OF THE EARTH'S ATMOSPHERE AT SEA LEVEL




If we take one cubic foot ($ft^3$) of air and weigh it, we find that it *weighs* 0.0736 $lbs/ft^3$ (1.18 $KG/M^3$).

Its density is:

$$\frac{0.0736\ lbs/ft^3}{32.1739\ ft/sec} = 0.002378 \text{ slugs}$$

while in SI the density is 1.18 $KG/M^3$.

The velocity of sound (c) in air for normal temperatures at sea level is dependent upon the density ($\rho$) of the air. (The temperature of the air has a major influence on its density.) The barometric pressure exerts a lesser effect under normal circumstances, the ratio of specific heats ($\lambda$) for air ($\lambda$ = 1.402) and the equilibrium gas pressure Ps = $1.013 \times 10^5\ N/M^2$ (the atmospheric pressure). The velocity (c) of sound can then be expressed as:

$$c = \sqrt{\frac{\lambda\ Ps}{\rho}}$$

The variable here is the density $\rho$ and density can be found accurately in specific cases by:

$$\rho^* = \left(\frac{0.00129\ H}{(1+(0.00367(^oK))76)}\right)10^3$$

* In $KG/M^3$

Where: H is the barometric pressure in cm of HG
$^oC$ is the temperature in Kelvins.

$9/5(^oC)+32 = {}^oF$ $\qquad$ $5/9(^oF)-32 = {}^oC$

When the amplitude of the air molecule being displaced is also known, then the sound pressure can be calculated as well:

$$P_{RMS} = 2\Pi fA\ \rho c$$

Where: $P_{RMS}$ is the root mean square sound pressure in pascals (pA) or ($N/M^2$)

A is the root mean square amplitude of the air molecule in meters (M)

$$A = 0.707\left(\frac{A\rho-\rho}{2}\right)$$

$\rho c$ is the density of air times the velocity of sound in air of that density. $\rho c$ is named RAYLS (406 RAYLS for air at sea level is the acoustic characteristic resistance)

Here we can note that the sound pressure is directly proportional to frequency, amplitude, density, and velocity. (Density and velocity, under normal measuring circumstances, become a constant.) Thus, as frequency changes the amplitude of the air molecules, displacement will vary inversely proportional to frequency :

$$1/A = \frac{2\Pi f\ \rho c}{P_{RMS}}$$

but directly in proportion to the pressure changes. This observation is borne out in watching loudspeaker cones exhibit larger and larger displacements in order to maintain the same $P_{RMS}$ as frequency is lowered. (Every time frequency is halved, amplitude is doubled if $P_{RMS}$ remains the same.)

Continued next page....

Since acoustic power is a function of the sound pressure squared:

$$Wa = \frac{(P_{RMS})^2(\text{area})}{\rho c}$$

Where: Wa is the total acoustic power in watts radiated by a source

Area is the total area through which the power passes at a given radius

We can now observe that every time frequency is halved and the cone's amplitude (displacement) remains constant, the power is reduced to 1/2 that of the higher frequency. Looked at another way, in order to maintain constant acoustic power a loudspeaker cone's amplitude (displacement) must double for every halving of frequency. (Equal Wa at all frequencies requires a constant $P_{RMS}$ at all frequencies.)

Finally, suppose we were to fully modulate ambient atmospheric pressure so that the -Δ atmos. was a vacuum and the +Δ atmos. was twice atmospheric pressure. We would then have a sound pressure *level* of:

$$L_\rho = 20 \log\left(\frac{1.013\text{x}10^5\rho a}{20 \text{ upa*}}\right) = 194 \text{ dB}$$

* 20 micropascals (20 upa) is the agreed upon reference value for converting sound pressures to sound pressure *levels* ($L_\rho$).

This earthly atmospheric environment leaves us with a substantial dynamic range media with inspirational side effects.

# WHAT WE FOUND WHEN WE MEASURED TELEPHONE LINE IMPEDANCES

HARRISON J. KLEIN, Engineering Manager at WIND in Chicago, has attended a couple of Syn-Aud-Con seminars. From a Newsletter item he built an impedance measuring system to study "dial up" telephone lines. His results, published in BM/E, March, 1981, and included with this issue, are eye openers in the use of the Kessler-Sprinkle approach for voltmeter impedance measurements. In the process, Mr. Klein added a most useful extension to the idea; namely, two bridging transformers for sampling current and voltage to ascertain which is leading and which is lagging in phase.

Harrison wrote us:

> "I thought you would be interested in the enclosed article, which reports some work we did at WIND using the voltmeter technique of complex impedance measurement described in a Syn-Aud-Con Newsletter some time ago. We have taken the results a step further by synthesizing hybrid balancing networks based on the measured impedance curves. On the 'well-behaved' lines discussed in the article, the results have been excellent. We're still trying to decide how to best deal with the 'troublesome' lines.
>
> "By the way, the second paragraph on page 72 originally said that the 'impedance of the unknown is then calculated using the equations in table 1.' For some reason the editors changed it to read 'using standard equations.' I would hardly call them 'standard;' in fact I find them to be quite ingenious."

Quoting from the article:

> "Notice first that each curve has areas of rising impedance, yet the phase angle always remains capacitive. This implies a transmission line effect that *cannot* be duplicated with simple lumped networks."

Mr. Klein goes on to describe the obvious difficulties this can cause in the design of balancing networks for hybrid transformers. One solution is to dial up another pair in the same trunk and use it as the balance network.

GARY MITCHELL tells the story of an experienced telephone engineer driving to work and observing a pair of overalls hanging across a pair of telephone open wire circuits. When he arrived at work, the younger engineers were busy using an impedance bridge to try to find the location of the fault. Our senior engineer bent over the bridge and finally muttered,"Look for a pair of overalls about four blocks from here on avenue X." Ever thereafter he was regarded with a degree of awe by the junior engineers.

Imagine an ETC measurement down the same line. Impedance measurements are basic but not always easy.

# USING THE NEW LOUDSPEAKER COVERAGE PLOTTING TECHNIQUES TO OBTAIN % ALcons

These are clever useful techniques when employed along with the concept of designing for the "farthest seat." The statement is made in Syn-Aud-Con classes that "if you design correctly for the farthest seat *and achieve even coverage*, you have successfully designed for all the seats within the coverage."

AN EXAMPLE

If you find that the farthest seat requires a Q of 10 at $D_{2SS}$, then you can adjust this $\%AL_{cons}$ as both the range (in dB according to inverse square law level changes) and the coverage angle (also in dB relative to some axis of reference).

If we convert these two levels, after algebraic addition, into a power ratio, which we will call K, then the $\%AL_{cons}$ at the point of observation becomes :

$$\%AL_{cons} \text{ at observer} = \left(\frac{\%AL_{cons} \text{ at } D_{2SS}}{K}\right)$$

Now suppose as we inspect our coverage we find a seat that is +3 dB closer than the range at $D_{2SS}$ but is on the -6 dB area of the coverage pattern. Then the $\%AL_{cons}$ for that seating area becomes:

$$\%AL_{cons} = \frac{10}{10^{\left(\frac{3-6}{10}\right)}} = 19.95\%$$

In the Altec version, a seat +3 dB closer is printed on their program as (-3 dB). Their coverage contours indicate correctly -3, -6, -9, etc.

Obviously, the name of the game is to keep range algebraically added to coverage as near zero as you can. The "N" factor still enters in as you find you need more than one source in order to maintain coverage and these equations then become:

$$\%AL_{cons} \text{ at observer} = \frac{(\%AL_{cons} \text{ at } D_{2SS})}{N \cdot K}$$

The inherent danger in these new programs lies in their tempting the designer to think that coverage is a substitute for the careful calculation of $Q_{min}$, $D_{2SS}$, N, and K.

The inherent virtue is that, for those sufficiently clever, they allow rapid evaluation of all these parameters over a wider calculating area with little extra effort.

# SPEECH PRIVACY DOCUMENTS

Sometimes it is difficult to determine where to order government documents. It often taxes perseverance. In Volume 8, No. 2 (page 27), we were able to give you the order information for the very valuable *Compendium of Materials*. Now we have been able to find the "Speech Privacy" documents, formerly available through the Superintendent of Documents.

We wrote:

General Services Office
U. S. Government
Washington, D.C. 20405

for the following material (it was sent free):

1. PBS-C.1 Test Method for Direct Measurement of Speech Privacy Potential Based on Subjective Judgments with.....

   A. Appendix No. 1 - Amendments to Test Method PBS-C.1 to include Evaluation of an Additional Capability for Fire Management Communication and Paging Messages.

2. PBS-C.2 Test Method of the Sufficient Verification of Speech Privacy Potential Based on Objective Measurements Including Methods for the Rating of Functional Interzone Attenuation and NC-Background with.....

   A. Appendix No. 2 - Amendments to Test Method PBS-C.2 to include Evaluation of an Additional Capability for Fire Management Communication and Paging Messages.

3. Guide for Acoustical Performance Specification of an Integrated Ceiling and Background System.

# JUNE 1981 CLASS

# DIGITAL QUOTES

Every month the evidence increases that digital recording is in more trouble than originally suspected. Here's Peter Sutheim again, this time describing a demonstration by Dick Heyser showing how to hear the digital "grunge." We suppose it doesn't really matter too much because a large number of us aren't buying digital and those who are will provide the needed "impartial" test. The only trouble with this type of testing is that a premature approach to the market place creates a "market memory" that interferes with a reintroduction for many years.

As we have declared from the beginning of this whole sordid affair, Syn-Aud-Con believes that digital recording more than likely contains the seeds of the future. More the regret if it's destroyed for this decade by greed-driven manufacturers lacking sufficient introspection to realize that what they see as a short-term profit is destructive of their long-term well-being.

Excerpts from *The Grounded Ear* by Peter E. Sutheim:

A simple experiment: run a sine wave test signal through a level-adjusting potentiometer into an analog-to-digital converter, using an appropriate anti-aliasing filter. Then reconvert into analog form and feed the signal to a power amplifier and speaker through an inverse level control, so that as you reduce the level of the input signal to the A/D converter, you automatically raise the level of the output signal, keeping the loudness of the tone constant.

Result: using a 440Hz tone (musical "middle A") into an 8-bit A/D with a 50kHz clock rate, the tone sounds nice and clean at first, with only a little background hiss to distinguish it from the tone straight out of the oscillator. As the input signal is reduced (remember, the loudness of the tone remains constant), the timbre of tone begins to change, first subtly and then radically. From the pure, sweet, dull character of the nearly perfect sine wave, the color changes through a slight edginess into nasality and finally into a nasal, irritating whine, something like a monotonous mosquito.

Hypothetical? Nope. Actual demonstration conducted by audio theoretician and writer Richard Heyser during a Sunday seminar called "New Wave Audio Measurements." I was there, and besides hearing the erstwhile sinusoid I got to see its waveshape disintegrate on an oscilloscope and its spectral footprint broaden on a spectrum analyzer.

Well, so it was an 8-bit system. But to me it seems plausible that such degradation at low levels exists also with 16-bit systems, though to a much smaller degree. It *is* a matter of degree: *how often does the degradation become audible*, assuming musically sensitive and experienced listeners? Not "does it happen?" (of course it happens), but "how much are we going to notice it?"

**Some days later** I attended a presentation by Dr. Thomas Stockham at which he played several examples of digital recordings made with his Soundstream system. The playback system set up for the occasion was poor; the sound was at no time beautiful and seldom pleasant, but that was not all attributable to Soundstream. What made me sit up suddenly, though, was a peculiar, alien quality to a soft, sustained violin tremolo background in Stravinsky's *Firebird*. The sound was less like bows on strings than like teeth gnashing. (While writing this, I listened to my copy of the Telarc release of the same performance and did *not* hear that objectionable sound.) After the presentation I asked Stockham whether he had heard what I had heard and whether he might attribute it to quantization errors in very low-level material. (Others in the audience heard it also.) His response was to deny categorically that audible quantization noise or distortion was occurring. Later, Richard Heyser, who was also present at the demonstration, assured me that what I had heard was a form of the "grunge" he had produced in the 440Hz sine-wave tone during his seminar.

**Meanwhile, designer/consultant** John Curl suggests, in a recent interview in *The Absolute Sound*, that the single most beneficial improvement in digital audio would be to double the sampling rate to 100kHz. This would not only move the anti-aliasing filter cutoffs much farther out of the way of the audio, but might even reduce the need for such absurdly sharp cutoffs. And of course the resolution of the system would be doubled.

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*THE AUDIO AMATEUR* 2/1981

From *dB Magazine* April, 1981; Excerpt from article by Barry Blesser:

BARRY BLESSER

# dB Digital Audio

## Clipping

### NON-HARMONIC DISTORTION

Clipping would appear to be a well-understood phenomenon. A clipped sine wave produces odd harmonics of the clipping is symmetric and all harmonics if unsymmetric. Similarly, it also produces extensive intermodulation distortion if more than one frequency is present. The interesting aspect of digital clipping is that the harmonics may beat with the sampling frequency and be moved to other frequencies. There are many ways in which we can demonstrate this. The digital word can only contain frequencies which are less than half the sampling frequency. What happens to the fifth harmonic of a 9 kHz sinewave in a 50 kHz sampling system? The answer is that it becomes 5 kHz, since 5 x 9 kHz is 5 kHz less than the 50 kHz sampling.

Yet another way to look at this is in terms of the anti-aliasing filter. This filter is to remove all components which are above half of the sampling frequency. Even if the filter is perfect, high frequencies could appear at the A/D converter if they are generated *after* the filter. Overloading the converter is like adding the illegal components and short-circuiting the function of the filter.

Clipping can thus be very bad if it is caused by steady-state high frequency signals. It is quite possible to create a case for audible distortion products leaving a digital audio system without any apparent input entering. The answer is that a very high frequency product beats with the sampling rate.

Continued next page.....

From *dB Magazine* March, 1981; Excerpt from article by Barry Blesser:

BARRY BLESSER

# dB Digital Audio

## Distortion

If you are keeping your wits about you, you may be asking: If the effect produces third harmonic distortion, and if we use high frequencies, shouldn't the harmonic component be filtered out by the filter? A test frequency of 18 kHz in a 50 kHz sampling system will produce a component at 3 x 18 kHz or 54 kHz. This is clearly outside the passband of the output filter. The answer is difficult to demonstrate, but the third harmonic will beat with the sampling frequency to produce a 4 kHz distortion since 54 is 4 kHz above 50 kHz. A 17 kHz sinewave will produce a 1 kHz distortion. The slew distortion is not harmonic but it becomes a low-frequency component. This would not be true for an 8 kHz input, since the third harmonic is 24 kHz which would be filtered out. Also, a 4 kHz input would produce a 12 kHz third harmonic, which would not be filtered nor would it beat with the sampling rate.

In terms of perception, 16.5 kHz would be very bad since it produces a 500 Hz distortion. The input signal is not audible but the distortion is clearly heard. No signal masking will cover the 500 Hz.

### TESTING

It should now be clear that the most difficult test for a digital audio system is a full-level signal at approximately one-third the sampling frequency. If the system performs well in this case, there is probably no distortion problem in the entire chain. It is a simple test and it does not require fancy equipment if you use your ears. Normally, one could not use an audible test since the distortion component was a multiple of the fundamental. The ear, as well as all the support monitoring equipment, generates more third harmonic than the equipment. However, in this special case, the distortion is moved to a new frequency which is very audible. None of the other equipment in the studio can move a single harmonic to a new frequency. Intermodulation tests are similar but they use two input signals to create the beats. In this case, the sampling frequency replaces one of the reference signals. ■

From *dB Magazine* May, 1981; Excerpt from article by Norman Crowhurst:

I venture to suggest, with some temerity, that John Diamond may be right and that, if we persist in ignoring his findings, we may be sorry in the end. What if we'd persisted in sticking to our theory that the refrigerator clicking in and out had nothing to do with hum in our audio system? But that was a little more obvious, we must admit.

Reproduced below is Dr. Diamond's latest attempt to obtain "equal time" for the unprecedented unilateral attack on his work by the *AES Journal*.

We always hope for the dissolving of adamant thought based on prejudice, fear, ignorance and other unscientific bases. Therefore, while we'd not place bets on it, we can lend a neutral stance to the *hope* that sometime the AES will recognize its responsibility to allow Dr. Diamond his day in court.

Dear Dr. Fehr: (Managing Editor, Journal of the AES)

I request that you publish these comments of mine on the apparent failures of Morgan and Clark to replicate my results on the stressful effect of listening to digital recordings:

1. There is a level of expertise required to carry out the test successfully. On an average day I perform the test some 350 to 500 times, and I have done this for many years. This is certainly more than the eleven or so "push tests" of Morgan or the few tests that Clark performed. One of the basic properties of music is that it increases strength on this testing. For Clark to find on several occasions that analog-produced classical music actually caused a weakening effect is an indication that his testing was faulty. (Incidentally, Clark refers to my "coaching" him. This alludes to a telephone conversation that we had. He declined to meet me personally in order for me to instruct him in carrying out the test correctly.)

2. Morgan found other parameters such as the EEG and GSR did not reveal the problem. Of this I am well aware. The problem is revealed, to date, only by the test I advocate. It also seems to be discernible by skilled listening, as I have received many reports from audio amateurs and professionals that they can discern with their ears the stress factor which the test indicates.

3. When I first presented my results at the AES meeting, and on every occasion since, I have offered to inform the digital recording manufacturers on ways to overcome the effect. Let me mention just one: I found, for reasons of which I am not aware, that increasing the length of lead between pickup and preamplifier or between preamplifier and amplifier would have the effect, under many circumstances, of overcoming the digital effect, as will altering the

Continued next page....

capacitance of the cable. Clark's introduction of an equalizer into the test situation will overcome the digital effect. In fact, this is the method I use to overcome it when listening for my own purposes. And of course the length of the leads of his comparator may also affect the results.

4. There are numerous variables which require control and which were neglected by Morgan and Clark. When and only when they attend to all the variables as I do in my research, and when they develop a level of expertise in performing the test, will it be possible to compare their results with mine.

Yours sincerely,

(signed) John Diamond, M.D.
Institute of Behavioral Kinesiology
Drawer 37
Valley Cottage, NY 10989

## AES Response

Robert O. Fehr, Editor of the *AES Journal*, not only refuses to publish Dr. Diamond's or Don Davis' replies to totally unsubstantiated attacks on Dr. Diamond, but further refuses to allow Don Davis to even see the alleged reviewer's comments. We'd be pleased to see Mr. Fehr's resignation as he is not an editor serving a peer society journal but is a man making unilateral editorial decisions based upon narrow and prejudiced considerations.

Shame!

Dear Mr. Davis: 1981 April 22

With reference to your letter of March 27, it is the editor's prerogative to make review comments available to the author verbatim or in summary. This is our practice. Following this practice, I sent you my summary of comments.

As I stated in my letter of April 25, my editorial decision is based on reading these reviews, and giving the whole matter due consideration.

Sincerely,

(signed) Robert O. Fehr
Editor, Journal
AES
60 East 42nd Street
New York, NY 10165

## A SYN-AUD-CON ENDORSEMENT FOR A CANDIDATE FOR THE AES BOARD OF GOVERNORS

Syn-Aud-Con's policy with regard to equipment or people is to inform you whenever we find a superlative new product or a remarkable individual's talent. We'd prefer to keep negative reporting to the barest minimum possible.

Glen attended the recent February Heyser TEF™ Workshop.

Thus, it is with real pleasure that we hear that GLEN BALLOU is a candidate for the Audio Engineering Society (AES) Board of Governors. We would be pleased to hear that those Syn-Aud-Con graduates who are AES members will support wholeheartedly Glen's election through their votes and recommendations to AES members that are not yet Syn-Aud-Con graduates.

The AES is at the threshold of either their own self-destruction through decreasing relevant contact with their membership or, through intelligent leadership of the caliber a Glen Ballou can provide, ready to enter into a new and more vital era wherein the limitations of the past, petty political, geographically provincial, and supportive of too narrow interests will be replaced by meaningful support of the entire audio industry.

Glen's work with Syn-Aud-Con, the editor of the revised *Audio Cyclopedia*, and as Papers Chairman of recent AES Conventions leaves us without a single doubt as to his capabilities, integrity and drive. We sincerely hope all graduates will join with Syn-Aud-Con in support of Glen.

# THE JUNE 1981 BCIT CLASS

# TEF™ ABSORPTION MEASUREMENTS

One of the most interesting effects we have been asked to measure with our TEF™ apparatus was brought to our attention by DOUG KENNEDY of Harford, Kennedy, Wakefield, Ltd. in the Canadian class.

Doug had a situation where it was desirable to use some very absorptive fiberglas insulation material which shed a good deal of particles in its natural state.

Because of these particles, it was determined they would wrap the fiberglas material in a thin mylar sheet, provided that the thin material didn't materially affect the required absorption (1 mil, 2 mil, and 3 mil thicknesses were being considered).

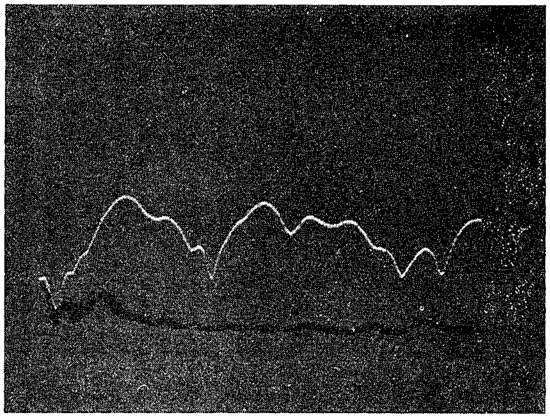


FIG. 1: The frequency response of the fiberglas without a covering

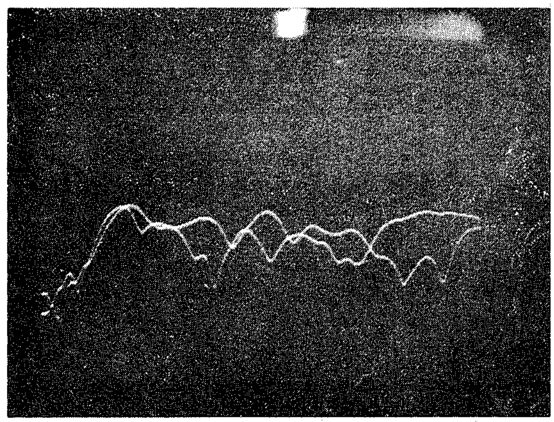


FIG. 2: The 1 mil cover

Much to all our surprise, even the 1 mil became a reflective surface above about 8 KHz.

Measurements of this type are particularly rapid and provide totally usable results because what is desired is the value of a change on a relative rather than an absolute scale.

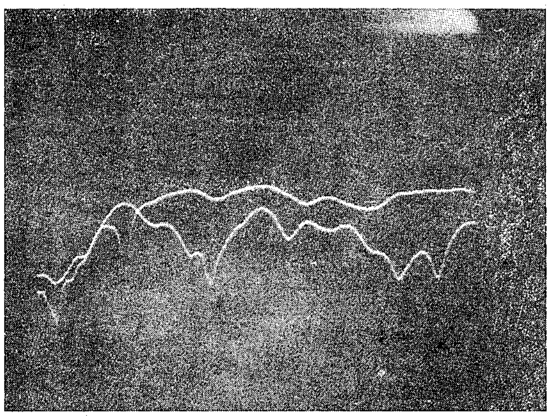


FIG. 3: The 2 mil cover

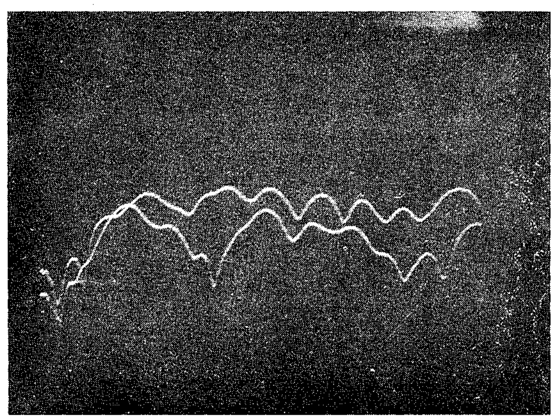


FIG. 4: The 3 mil cover (the ripple is caused by the cover not being stretched tightly)

FIG. 5: Ken Wahrenbrock and Doug Kennedy applying the thin mylar sheets to the surface of the fiberglas test material.

After reviewing this data the thought occurred to us that the effect might indeed have useful application in any situation where you deliberately wished to vary selected areas of, for example, Sonex absorption on a frequency selective basis. We believe thin membranes of this sort might lend themselves to such applications with a minimum of expense.

Whenever you observe an effect (sometimes detrimental to what you originally had in mind) don't forget to consider other applications where its seeming fault becomes its virtue.

# VOLTAGE AMPLIFICATION vs. INSERTION GAIN

Voltage amplification (A) is defined as: $A = 20 \log\left(\frac{E_{out}}{E_{in}}\right)$

and is very frequently *mislabeled* as "Gain." Such mislabeling can cause serious results if the figure is included as part of a "system" level graph.

Since the desired end result for any device within a system is to specify what happens to a sound level meter's reading out in the audience area due to the "insertion" of said device into the system, we don't want the voltage amplification but rather the "insertion gain" (G).

$$G = 20 \log\left(\frac{E_{out}}{E_{in}}\right) + 20 \log\left(\frac{R_{in}}{R_S+R_{in}}\right) + 10 \log\left(\frac{R_S}{R_L}\right) + 6.02 \text{ dB}$$

Where:

G is the insertion gain in decibels

$R_{in}$ is the input resistance in ohms of the device

$R_S$ is the source resistance of the source driving the device

$R_L$ is the load resistance across the output of the device

$20 \log\left(\frac{E_{out}}{E_{in}}\right)$ is the *"voltage amplification"*

$20 \log\left(\frac{R_{in}}{R_S+R_{in}}\right)$ is the *"coupling factor"*

$10 \log\left(\frac{R_S}{R_L}\right)$ is the *"impedance mismatch"*

+ 6.02 dB is the *difference between an open circuit and a matched circuit's level*

EXAMPLE NO. 1

Suppose we are measuring a mixer amplifier. $R_S$ is usually a microphone connected to its input so $R_S = 150\Omega$. It has a quoted output of 600Ω (though higher impedances may well be connected to it) so $R_L = 600\Omega$ is used for calculation purposes. In a well designed mixer of this type, $R_{in}$'s of from 1500 to 3,000Ω are common in order to gain a signal-to-noise advantage and make the effect of microphone impedance variations negligible. Let's say in this case it is 1500Ω. If we have an output (open circuit) from our microphone of 0.98 MV and we find that at the output of the mixer we have 21.8 volts, we can then calculate the "insertion gain" for the mixer.

$$20 \log\left(\frac{21.8V}{.00098V}\right) = 86.94 \text{ dB}$$

$$20 \log\left(\frac{1500}{150 + 1500}\right) = -0.83 \text{ dB}$$

$$10 \log\left(\frac{150}{600}\right) = -6.02 \text{ dB}$$

$$\underline{+6.02 \text{ dB}}$$

$$\text{insertion gain} = 86.11 \text{ dB}$$

Therefore, in this case (A) and (G) are approximately the same.

EXAMPLE NO. 2

Here we have a low impedance source $R_S = 150\Omega$ connected to a high impedance input $R_{in} = 10{,}000\Omega$ and an output resistance of $R_L = 8\Omega$.

$$E_{in} = 1.0V \quad \text{and} \quad E_{out} = 0.5V$$

Once again, we calculated the *insertion* gain

$$20 \log\left(\frac{E_{out}}{E_{in}}\right) = -6.02 \text{ dB}$$

$$20 \log\left(\frac{10{,}000}{150 + 10{,}000}\right) = -0.13 \text{ dB}$$

$$10 \log\left(\frac{150}{8}\right) = 12.73 \text{ dB}$$

$$\underline{+6.02 \text{ dB}}$$

$$\text{insertion gain} = 12.6 \text{ dB}$$

Continued next page....

VOLTAGE AMPLIFICATION VS INSERTION GAIN continued....

In this case, A = -6.02 dB (a loss) and G = +12.6 dB (a gain). The difference of over 18 dB could be a devastating experience if you mistakenly relied on (A) instead of (G) as a predictor of what inserting this device into the system would cause as a *level* change in the system's output.

Again, let's emphasis that we are not criticizing the practice of "circuit" engineers who think in terms of "voltage swings" as they design discrete circuits. It is the "system" designer who must, at the output, supply the *acoustic power* to each listener's ears if the system is to work successfully, and he must know what the *insertion gain* is if he is to know in advance how each component will affect his system.

## SYN-AUD-CON HOME ON THE ROAD

Syn-Aud-Con has a new mode of transportation for the Fall 1981 traveling classes.

It is a 35 foot Foretravel Motor Home. Motive power is a Caterpiller (sometimes called Clatter-piller) 636 CID V-8 model 3208 diesel with an Allison AT-540 transmission in a "pusher" (rear engine) configuration. The basic chassis is an Oshkosh engineered specifically for such heavy duty service.

The picture was taken by KEN WAHRENBROCK as we were parked by the Columbia River in the state of Washington. Our Dodge van (driven by Ken) carried the equipment for an "on the road" class in Vancouver, British Columbia, and serves as a good scale to measure the size of the motor home.

We will be "on the road" in this new "home" from late July to early December bringing Syn-Aud-Con classes to the Midwest and East again after a two year building program at the West Coast Seminar Center.

## BOOKS OF INTEREST

Buckminster Fuller, 85 years of age in July, 1980, having consumed over 1,000 tons of food, water and air with 146 pounds remaining, has written another totally fascinating book entitled *Critical Path.*

A typical Fuller utterance, "Cosmically acceptable and effective decisions of humanity regarding such matters (human survival) will not be made by leaders single or plural, political or religious, military or mystic, by coercion or mob psychology." "The effective decisions can only be made by the independently thinking and adequately informed human *individuals* and their telepathetically intercommunicated wisdom--the wisdom of the majority of all such human individuals--qualifying for continuance in universe as local cosmic problem-solvers--in love with the truth and in individually spontaneous self commitment to absolute faith in the wisdom, integrity, and love of God, who seems to wish Earthian humans to survive."

And on the aptly named "generation gap" of the 1960's, he writes: "The university and college students who became the first to make the world news as dissidents in 1965 and 1966 were born in the years TV came into the American home." "The class of 66'ers were the first human beings to be reared by the "third parent," whose TV voice and TV presence were often heard and felt by the children much more than those of the two blood parents......With TV making it clear to the young that the parents did not know much about anything and were not 'the authority,' the young responding to intuition, said to themselves, 'I am going to have to do my own thinking and take my own actions.'" "Nonetheless, they were utterly unskilled in world affairs, highly idealistic, and easily exploitable."

And frightening in its implications if the premise is accepted:

"America is utterly bankrupt externally in terms of balance of trade due to *its own* oil companies now operating as Arabian business......The USA cannot even pay the annual interest on its 800 billion dollar national debt."

The basic problem with housing is contained in a sentence as a footnote, "In 1978 over one billion dollars was spent just for transferring home ownership deeds."

This book is supremely worthwhile if you read only its Chapter Three, "Legally Piggely."

And finally, some hope in the confusion. While Murphy's Law says that the light at the end of the tunnel is an oncoming locomotive, Fuller points out that "Such individuals as Pythagoras and Buddha were *unanticipated* by the behind the scenes physical power structures." "They were probably unnoticed at first."

Never boring, often inspired, not necessarily someone you'll agree with but a powerful catalyst to the thinking process, Buckminster Fuller continues to outrage the ingrown, humble the intellectual pretense, and inform the receptive.

# CLASSIC COMB FILTER EQUATIONS

Olson and Massa's classic book, *APPLIED ACOUSTICS*, published by Blakeston in 1934, contained the following description of how comb filters are generated:

In using the equations remember that integers are whole numbers such as 1, 2, 3, etc. Thus, when the distances of $\sqrt{d_1^2+4h^2} - d_1$ are integer multiples of one wavelength, the signals are in phase.

$$N\lambda = 1\lambda,\ 2\lambda,\ 3\lambda,\ \text{etc.}$$

When the distance difference between the two

are: $\frac{2N-1}{2}\lambda$

or: $\frac{2x1-1}{2} = .5\lambda$

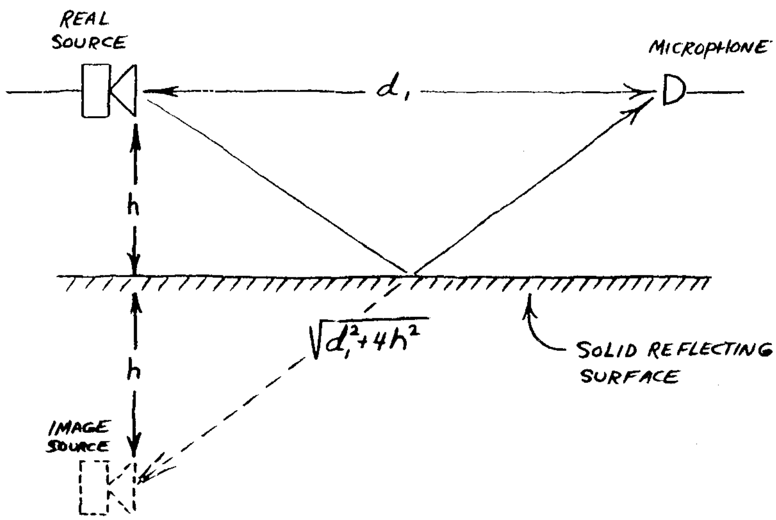
$\frac{2x2-1}{2}\ \frac{3}{2} = 1.5\lambda$

$\frac{2x3-1}{2} = \frac{5}{2} = 2.5\lambda$

the two signals are "out of phase."

The illustration clearly shows the "model" employed to obtain the equations.

REFLECTION GENERATED "COMB FILTERS"




Out of phase $\sqrt{d_1^2+4h^2} - d_1 = \frac{2N-1}{2}\lambda$

In phase $\sqrt{d_1^2+4h^2} - d_1 = N\lambda$

Where: N is an integer and $\lambda$ is the wavelength

## CALCULATING ANOMOLIES AND PHASE ANGLES

When two signals arrive at a common observation point at slightly different times (distances), we can observe, as if both signals originated from the same source, their *relative* path differences (not frequency dependent) and *relative* phase angles (frequency dependent).

Such signals will be: In phase every $\left(\frac{\text{Vel of sound in ft/sec}}{\text{path difference in ft}}\right)$

If two signals arrive 1.13 feet apart then: $\frac{1130\ \text{ft/sec}}{1.13\ \text{ft}} = 1000\ \text{Hz}$

and these two signals will be in phase every 1000 Hz so that in phase is 1000, 2000, 3000, 4000 Hz, etc.

These same two signals will be out of phase every: $0.5\left(\frac{\text{Vel of sound in ft/sec}}{\text{path difference in ft}}\right)$

so that: $0.5\left(\frac{1130}{1.13}\right) = 500\ \text{Hz}$

and the two signals will be out of phase at 500 Hz, 1500 Hz, 2500 Hz, 3500 Hz, etc.

Note that the spacing between out of phase frequencies remains the same as the spacing between in phase frequencies (in this case 1000 Hz).

## THE PHASE ANGLE

At the "in phase" frequencies the relative phase angle is $0^\circ$. At the "out of phase" frequencies the relative phase angle is $180^\circ$. The phase at any frequency difference in between ($\Delta f$)* can be found by:

$$\text{phase angle for } \Delta f = \Delta f\left(\frac{180^\circ}{\text{in phase frequency-out of phase frequency}}\right)$$

For instance, imagine a situation where the "in phase" frequencies are 1000 Hz, 2000 Hz, 3000 Hz, etc., and the out of phase frequencies are 500 Hz, 1500 Hz, 2500 Hz, etc., and we wish to know the relative phase angle between the two signals at 1250 Hz.

$$\text{phase angle at } \Delta f = 250\left(\frac{180^\circ}{500\ \text{Hz}}\right) = 90^\circ$$

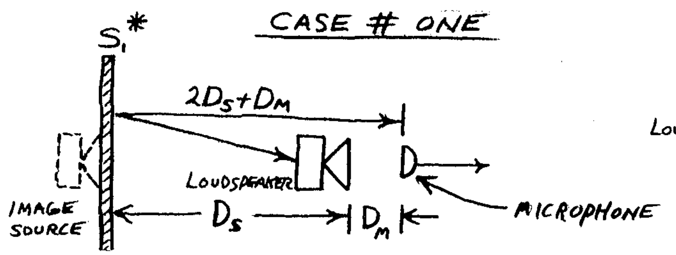
*$\Delta f$ is the difference between the frequency of interest and the in phase frequency.

Continued next page.....

CLASSIC "COMB FILTER" EQUATIONS continued

When knowledge of the individual levels of two signals and their relative phase angles are available, it is easy to plot their resultant frequency response. The design of an LEDE™ control room's diffuse rear wall depends upon the generation of sets of anomolies that have the "peaks" of one set covering the "nulls" of another set all occuring within the primary Haas time zone.

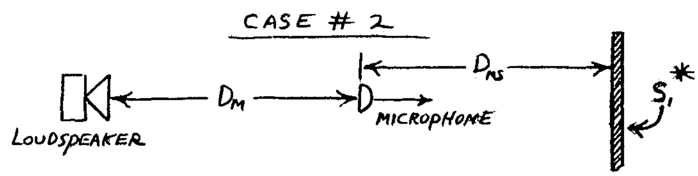
CALCULATING RELATIVE LEVELS OF REFLECTIONS




Influence of surface $S_1$ on measured signal at microphone equals:

Reflected signals relative level =

$$20 \log\left(\frac{D_m}{2D_s+D_m}\right)$$




Influence of surface $S_1$ on measured signal at microphone equals:

Reflected signals relative level =

$$20 \log\left(\frac{D_m}{2D_{ms}+D_m}\right)$$

Where "$S_1$" is absorptive then the equation becomes:

Reflected signal level =

$$20 \log\left(\frac{D_m}{2D_{ms}+D_m}\right) + 10 \log (1-a)$$

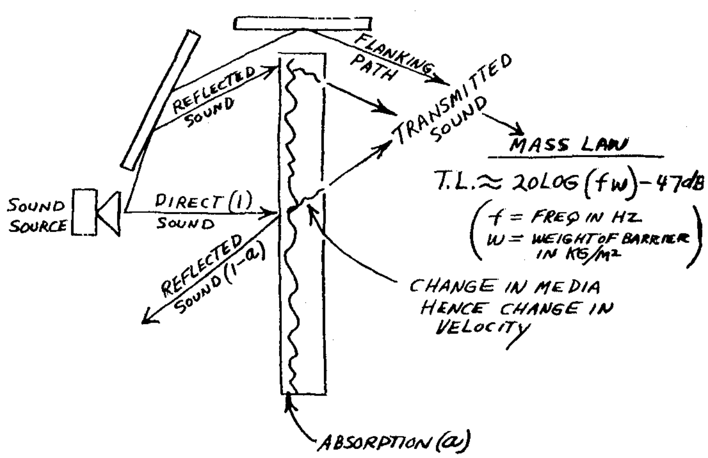
In the case of substantial transmission loss then these losses can be added as required.

$$\text{T.L.} \approx 20 \log (fw) - 47 \text{ dB}$$

(See above illustrations for explanations of symbols.)

* Assuming $S_1$ is non-absorptive and non-diffusive.

ABSORPTION, REFLECTION AND TRANSMISSION OF BOUNDARY SURFACE AREAS




$$a = 1 - 10^{\left(\frac{-dB}{10}\right)}$$

$$dB = 10 \log (1-a)$$

# ARTICLES OF INTEREST

A recent edition of "Electronics" magazine carried a writeup on the use of sound wave velocities as a way to accurately measure very high temperatures. Now you can use your ETC to tell the temperature.

In the case mentioned above, four years of research at Sandia National Laboratories in New Mexico resulted in a new ultrasonic thermometer capable of accurately measuring temperatures up to 2845°C with a ± 1° accuracy. See page 132 of the June 30, 1981, "Electronics" for further details.

Continued next page....

ARTICLES OF INTEREST continued....

An article entitled "Zen and the Art of Using Wireless Microphones" written by Dale Scott of H M Electronics for the April 1981 issue of *REP* magazine contains much practical information on the not always easy to use wireless systems. In Syn-Aud-Con's, albeit relatively limited, experience with wireless microphone systems we have used only two systems very successfully--HME and Swintek.

Dale Scott covers everything from very specific reference to preferred batteries (Mallory Duracell MN 1604) to why one user orders condoms by the gross without telling purchasing why (performer perspiration protection). Many unusual applications are discussed with a degree of frankness about inherent difficulties that is refreshing.

Interestingly, when we poll Syn-Aud-Con classes regarding their experiences with suitable wireless microphone systems, we find our favorites always head the list. This is not to say that many graduates have not had favorable results with other brands but merely that a sizeable majority have had their best results where we have.

Whenever we run into a temporarily unsolvable wireless microphone system interference (such as trying to use Canadian hydrological frequencies on the Canadian seacoast), we are always appalled at the loss of freedom being hooked to the microphone cable implies. Top quality wireless systems are very habit forming.

We suggest you take a look at this article and its practical, useful sharing of ideas. You may be the one to creatively extend one of the unusual applications described.

Part II of "The Start of Something Big" by John T. Mullins written for *THE MIX* magazine appeared in their April 1981 issue starting on page 30.

This exciting continuation of John Mullins' recollections from the genesis of tape recording indicates John's ability to back the right people and his courage in doing so at particularly crucial moments when he opined that Ampex (Harold Lindsey in particular) would succeed in building a machine that recorded as well as it played back for Jim Middlebrook of ABC. The result of this act of faith was that Ampex received a *firm* order for 12 of the original Ampex Model 200's.

The inception of magnetic recording telemetry, video recording (still a mixed blessing), and the continuing development of professional audio machines are recorded here in a way serious historians of the recording arts will one day consider extremely valuable source material.

DAVID L. KLEPPER, William J. Cavanaugh and L. Gerald Marshall tackle the frequently overlooked problem of "Noise Control in Music Teaching Facilities" in the Sept.-Oct., 1980, issue of "Noise Control Engineering."

The article is straightforward and contains many practical discussions of actual construction practices necessary to the accomplishment of noise control goals. For those of you with access to "Noise Control Engineering," we recommend you read this useful article.

The references listed are worthwhile also. I particularly looked to see if a publication was listed that we reviewed in January, 1976. I don't know if it is still available but, if you're interested, it's worth checking: "Planning and Equipping Facilities" from Music Educators National Conference, 1902 Association Drive, Reston, CA 22091. Request 32109948. Sold for $12 in 1976.

# CLASSIFIED

WANTED: HP41C with memory. Contact Frank Huang, 3141 East 62nd Avenue, Vancouver, B.C. CANADA V5S 2G6 604/438-1355.

TRADE: Like new UREI 537 Graphics for UREI 539 Room Equalizers. Contact P. Russell 212/736-5727.



"SYN-AUD-CON"

## SYN-AUD-CON SPONSORS

Syn-Aud-Con receives tangible support from the audio industry, and ten manufacturing firms presently help underwrite the expense of providing 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 graduate 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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