"SYN-AUD-CON"

SYNERGETIC AUDIO CONCEPTS

P.O. BOX 1134, TUSTIN, CALIFORNIA 92680

newsletter

VOLUME 5, NUMBER 2
JANUARY, 1978
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EDITORS: Don & Carolyn Davis

SYNERGETIC
Working together; co-operating, co-operative

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

EXCHANGE OF IDEAS

I met a man with a dollar
We exchanged dollars
I still had a dollar

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

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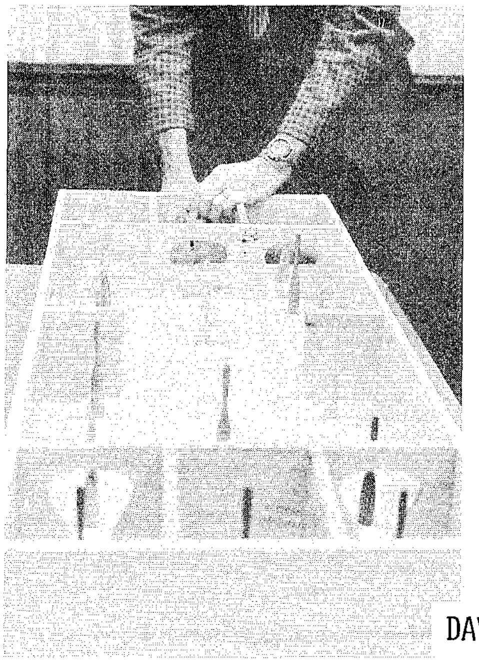
SYNERGETIC AUDIO CONCEPTS

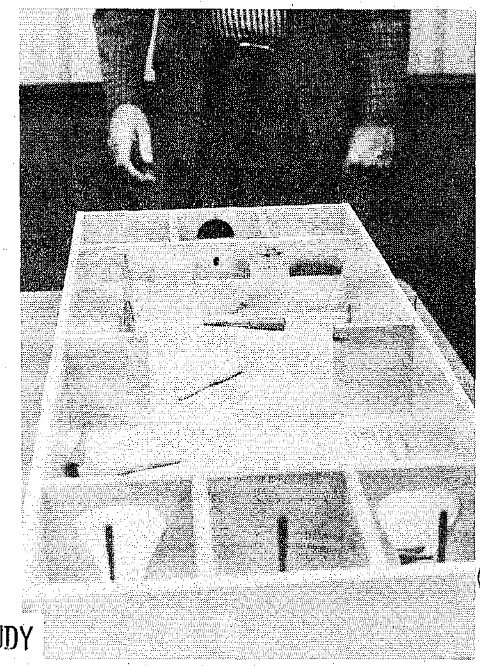
1978 SEMINAR SCHEDULE

Anaheim, CA - January 31-February 2
Seattle, WA - February 14-16
San Francisco, CA - February 22-24
Vancouver, BC - April 18-20
Los Angeles, CA - May 10-12
Chicago, IL - Mid-September
New York, NY - Late September
Washington, DC - Early October
Atlanta, GA - Mid-October
Orlando, FL - Late October
Houston, TX - Early November

SKITTLES

During our travels through Eastern Kentucky we came across a game called Skittles. Built by the students at Berea College, it is an early day Appalachian pinball game. A string is wrapped around a skittle (spinner) and pulled through a slot in the end board to start it spinning. From then on you have a superb demonstration of random processes as the skittle "walks" through the various rooms either knocking over pins that add to your score or pins that subtract from your score. So far the highest score in class (we've had it in the Nashville and Orlando classes) is 255 out of a possible 275.





DAVID CLARK HEARING PROTECTORS FOR JUDY

Judy has been known to depart the scene immediately, without regard for the distance home, upon being subjected to a loud impulse sound. A 458 Winchester magnum express rifle or a 44 magnum revolver causes Judy instant distress. Since I use David Clark hearing protectors it was only natural to try one of the lightweight models on Judy. I wear the large protectors during shooting, the Model 19A, and I have rarely encountered any device that performs as satisfactorily as this one.

Judy, with Barbara Cable, in the Orlando class wearing hearing protectors during the "tuning" session. Judy is modeling the David Clark model E-320.

A VERY SPECIAL GRADUATE SEMINAR, APRIL 30, 1978

RICHARD C. HEYSER
JOHN HILLIARD
JAMES MOIR
VICTOR PEUTZ

If you have yearned to meet, work with, and share experiences with the truly talented investigators and innovators of our audio world and have dreamed of discussing with such men their current ideas, experiments, and insights, then this special Syn-Aud-Con graduate seminar is for you.

There will be a one day gathering of Syn-Aud-Con graduates on April 30, 1978 - the Sunday before the Spring AES Convention in Los Angeles. We apologize for the Sunday date but as Monday, May 1st is the set-up day for AES this left us little choice. Lunch and dinner is included in the $150 fee. The meeting starts at 9:00 a.m. and concludes at 9:00 p.m.

The seminar will consist of special papers, panel discussions, question and answer sessions, and special demonstrations conducted by a most distinguished panel of experts.

RICHARD C. HEYSER

Dick is the inventor of Time Delay Spectrometry, TDS; a Fellow of the AES *and* ASA. He knows more about the testing of transducers than any other audio authority today. Through Dick's work the importance of minimum phase response in loudspeakers was brought to public attention and the philosophical basis of Time Align[R] was developed.

JOHN HILLIARD

John is a Fellow of the SMPTE, AES, ASA and IEEE. He is the engineer responsible for the development of the modern two-way loudspeaker system. He authored the first paper written on phase distortion. John was a close associate of James B. Lansing, Harvey Fletcher, Vern Knudsen and many, many other giants of the audio-acoustic industry. He designed and built the first intermodulation distortion analyzer. He was the first to write an article on the effect of Time Align[R] in loudspeakers and most impressive of all, he is providing leadership in the acoustical consulting field.

JAMES MOIR

Jim is the foremost acoustical consultant in England today. His early acoustical work included the design of over 40 movie studios and theatres. He was the first in audio history to make pulse tests in auditoria and correctly interpret the results. Jim was one of the very select group of engineers and scientists that brought British radar to readiness in time to meet the German attack which history has ennobled as the Battle of Britian. His latest in a long line of creative innovations is the Bruel and Kjaer type 4205 sound power source for directly measuring the acoustic power output of a sound source.

VICTOR M.A. PEUTZ

Victor is Europe's leading acoustical consultant. His consulting activity takes him worldwide. He is the acoustical consultant for the IRCAM project in Paris, a four-story underground acoustical music research laboratory. Victor is best known to Syn-Aud-con graduates for his remarkably useful work on the calculations of articulation loss of consonants in speech. Victor's work on when and how a reverberant sound field is established is fundamental, advanced, and as always with his work, immediately practical. He currently is at work on a rating system for music similar in nature to his speech intelligibility work.

****

PLEASE - GRADUATES ONLY

DETACH AND MAIL THIS PORTION

See Reverse Side for Registration Details

SYNERGETIC AUDIO CONCEPTS
P. O. BOX 1134, TUSTIN, CA 92680

DON: PLEASE REGISTER ME IN THE VERY SPECIAL GRADUATE SEMINAR, APRIL 30, 1978

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Check may be made to Syn-Aud-Con

☐ Check enclosed ☐ Charge to my MASTER CHARGE, BANKAMERICARD, VISA (*Circle one*)

☐ Deposit Enclosed ($50.) ☐ Payment in Full Enclosed ($150.)

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A VERY SPECIAL GRADUATE SEMINAR, APRIL 30, 1978, CONTINUED

I'd like to stress that we have been able to list here only an infinitesmal number of the accomplishments, talents, and current interests of these men. Their backgrounds probe deeply into every facet of recording, reproduction, reinforcement, and research both acoustic and electronic. These are *systems* experts in every sense of the word and not only represent the best of our past accomplishments in audio but are the true leaders in our present advancements. Best of all, these men are genuinely communicative. They know the value of a shared idea. They know audio intimately and are articulate.

This panel of mental megatonnage will be assisted by Cecil Cable and me. While the one-day meeting is open to Syn-Aud-Con graduates only, we will invite people who have made outstanding contributions to our industry, and our Syn-Aud-Con sponsors, which include many of our industry leaders.

We truly feel that we have brought together four audio "Wan Kanobios" and we believe that the focus of their ideas in a meeting of this type should generate an explosion of new work by all of the privileged participants.

Each participant will receive from Syn-Aud-Con a symbol of this meeting of the minds to identify them during the AES convention.

The meeting will be held in or near downtown Los Angeles. An Application is attached for you to fill out and return. If the attendance becomes too great we will have to limit attendance according to the date we receive your applications.

****

REGISTRATION AND FEES

ENROLLMENT INFORMATION: Early enrollment is suggested. To register for the session, complete and return the coupon on the reverse side. We will confirm your registration. You may send a $50 deposit and be billed for the balance or charged on Master Charge or bankamericard. The full fee is payable a week in advance of seminar date. Confirmed registrations cancelled up to five (5) days prior to the seminar date are eligible for full credit. Cancellations received less than 5 days prior to the seminar are subject to $50 administration charge.

SEMINAR HOURS: Session starts at 9:00 a.m. and closes at 9:00 p.m.

HOTEL ACCOMMODATIONS: A block of rooms will be reserved for participants at the hotel where the seminar is held until two weeks before the seminar starts. Hotel room, meals, (except lunch, dinner, and coffee breaks during seminar), and other hotel charges are the responsibility of the individual participant. A no-host bar will be available before dinner.

REGISTRATION FEE: The full fee is payable in advance

Each participant $150.00

PEUTZ
HEYSER
HILLIARD
MOIR

HEYSER
HILLIARD
MOIR
PEUTZ

Register Now

A VERY SPECIAL GRADUATE SEMINAR
APRIL 30, 1978

HILLIARD
MOIR
PEUTZ
HEYSER

MOIR
PEUTZ
HEYSER
HILLIARD

(Use Registration Form on Reverse Side)

BEARSVILLE RECORDING STUDIO

TED ROTHSTEIN, chief engineer at Bearsville Studio, JOHN STORYK, architect-designer of Sugarloaf View, Inc., of New York and Bearsville, and AL FEIERSTEIN, designer of the eclectic Acoustilog Reverb Timer, attended Update class in New York last Fall. They invited the Cecil Cable's and Carolyn and I to visit the Bearsville Studio on our way from the Boston to Philly class. (It's an interesting place to tow a travel trailer into but that's a separate story. Someday I'll write the Syn-Aud-Con 3T's - Travel Trailer Traumas.)

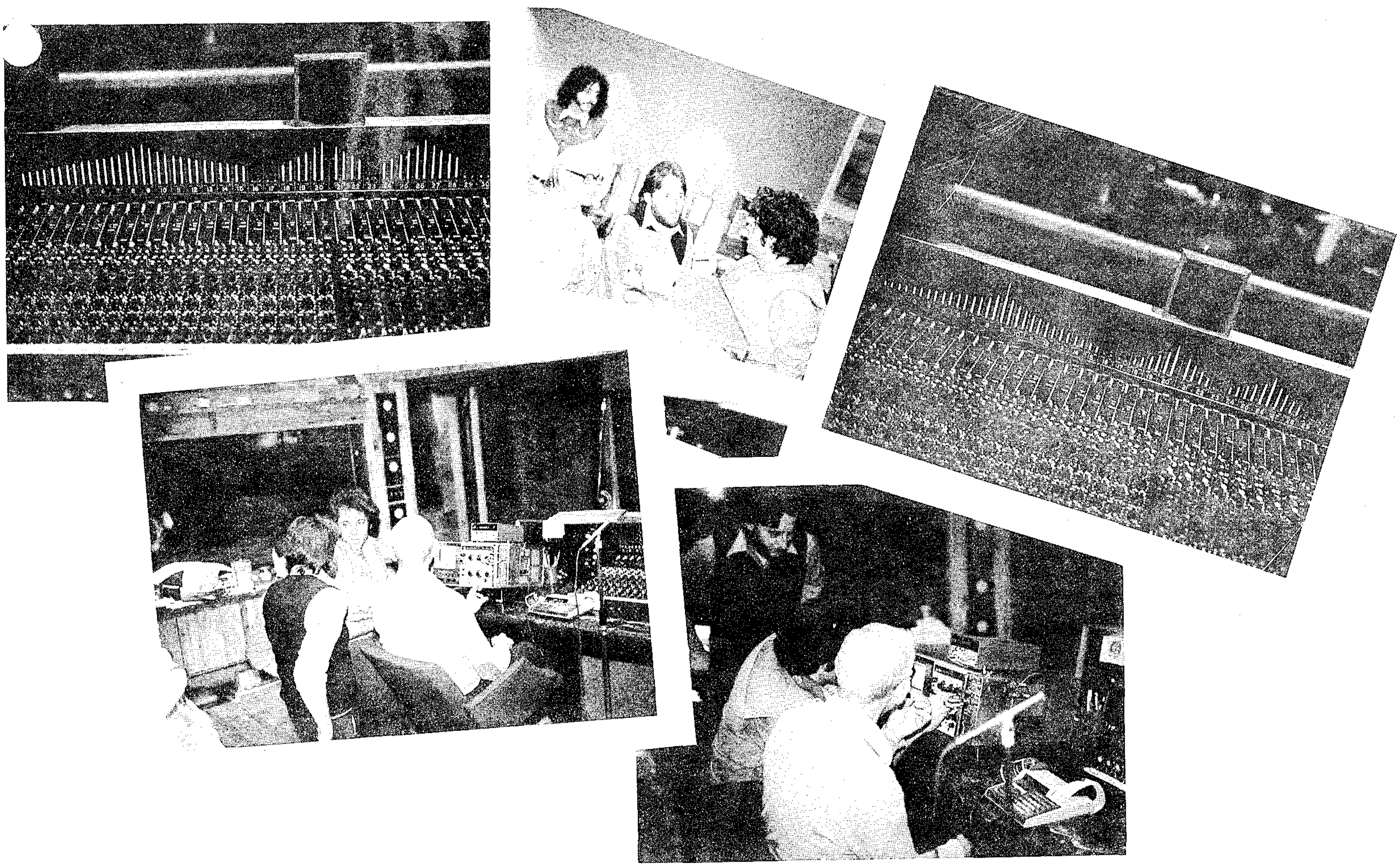
The owner of Bearsville is inadequately described as a Renaissance Man but it does begin to give at least a partial picture. Al Grossman has brought together at Bearsville a most interesting concentration of engineers, craftsmen and artists. His restaurant-theater projects alone are worth a visit to this unique community.

At Bearsville Studio we saw a really deluxe recording console originally manufactured by Quad Eight and creatively modified by Ted Rothstein - triple concentric equalizer controls, special metering, etc.

It is the special level indicators that particularly intriqued us. One can watch peak values in a context where such readings are really meaningful. Ted has asked that we not reveal the supplier of these displays as yet but as the photographs show, they are vertical line indicators.

The indicators can:

A. Read true RMS
B. Read true Peak
C. Read frequency
D. Hold on highest reading
E. Read with slow ballistics
F. Allow simultaneous RMS and Peak readings on the same indicator by using two intensities on the display device
G. Convert into reading 1/3-octave band levels where an analyzer is desired
H. Convert into *two* one-octave real time analyzers for studying the spectrum of program material for stereo playback
I. Read either the output from the console or the output from the tape recorder or compare them by displaying both at differing intensities.



BEARSVILLE, continued

The photographs on the previous page show the unusual level indicators. The darker photo allows the Peak readings (lower intensity) to be seen. The small loudspeaker on the console is a single con E^3 monitor that includes builtin compromise equalization to adjust for optimum speaker response at the mixer's position.

Cecil was able to demonstrate to everyone's satisfaction, using TDS, that the control room where this console was located *did not* develop a reverberant sound field. (Nor have we measured a control room that does.) This makes the study of early reflections vital (there are no late ones). The most satisfying and exciting aspect of measurement work in a control room is that the source of every anomoly can be quickly identified and almost as easily corrected.

Studios, restaurants, mobile recording vans, electronic labs, pleasant rural homesites, all deep in Rip Van Vinkle's woods, plus a new experimental theater for "live" acts of every description in a super supper club atmosphere left us with gratitude that we had been invited to be guests at Bearsville.

We expect to hear many future innovations out of this unusual technical group inasmuch as the talent, the opportunities, and the patron are all in sync.

COMPUTER PROGRAM TO CALCULATE RT_{60}

JIM HAWKINS, owner of Electro Acoustic Systems in Athens, GA and two-time graduate, gave us a beautiful program for the HP 67-97. Jim's program allows the HP 67/97 user to greatly speed up the calculation of RT_{60} from architects' drawings.

The program calculates the RT_{60} for six frequencies (the normal number provided for in acoustical absorption tables). After entering the basic program in the calculator in the normal way and initializing by pressing "A", a second magnetic card is used for each different type of acoustic material being considered.

After a short time it should be possible to develop quite a collection of standard material coefficients on these cards ready for use whenever the materials are encountered.

```
001  *LBLA     21 11
002   ST+9  35-55 09
003   STOA     35 11
004      6        06
005   STO7     35 07
006      1        01
007   ST+8  35-55 08
008  *LBL1     21 01
009   RCL7     36 07
010   STOI     35 46
011   RCLi     36 45
012   RCLA     36 11
013      ×       -35
014      1        01
015      0        00
016   RCLI     36 46
017      +       -55
018   STOI     35 46
019    X⇄Y       -41
020   ST+i  35-55 45
021      7        07
022   STOI     35 46
023   DSZi  16 25 45
024   GTO1     22 01
025      6        06
026   STOI     35 46
027   RCL8     36 08
028    RTN        24
029  *LBLB     21 12
030    SPC     16-11
031   STOD     35 14
032      6        06
033   STOI     35 46
034    P⇄S     16-51
035  *LBL2     21 02
036      .       -62
037      0        00
038      4        04
039      9        09
040   RCLD     36 14
041      ×       -35
042   RCLi     36 45
043      ÷       -24
044   PRTX       -14
045   DSZI  16 25 46
046   GTO2     22 02
047    P⇄S     16-51
048    RTN        24
049    R/S        51
```

NORMAL STORAGE		
	0.00	0
$\bar{a}$ 125 HZ	0.12	1
$\bar{a}$ 250 HZ	0.14	2
$\bar{a}$ 500 HZ	0.15	3
$\bar{a}$ 1000 HZ	0.16	4
$\bar{a}$ 2000 HZ	0.18	5
$\bar{a}$ 4000 HZ	0.19	6
	0.00	7
NUMBER OF MATERIALS	2.00	8
TOTAL S$\bar{a}$	42500.00	9
S$\bar{a}$ THIS ENTRY	42500.00	A
	0.00	B
	0.00	C
VOLUME	500000.00	D
	0.00	E
MERGE #	6.00	I

PROTECTED STORAGE		
	0.00	0
S$\bar{a}$ 125 HZ	5100.00	1
S$\bar{a}$ 250 HZ	5950.00	2
S$\bar{a}$ 500 HZ	6375.00	3
S$\bar{a}$ 1000 HZ	6800.00	4
S$\bar{a}$ 2000 HZ	7650.00	5
S$\bar{a}$ 4000 HZ	8075.00	6
	0.00	7
	0.00	8
	0.00	9
TOTAL S$\bar{a}$	42500.00	A
	0.00	B
	0.00	C
VOLUME	500000.00	D
	0.00	E
MERGE #	0.00	I

RT_{60}		
4000 HZ	3.03	***
2000 HZ	3.20	***
1000 HZ	3.60	***
500 HZ	3.84	***
250 HZ	4.12	***
125 HZ	4.60	***

The data card is (f) merge loaded, and the appropriate area is keyed into the calculator. "A" is then pressed. The calculator computes the number of sabins for each of the six frequencies and accumulates this information in the secondary (protected) registers. A running total of the surface area accounted for is maintained in Reg. 9, and a count of the number of different materials entered is displayed.

When all materials have been fed in, key in the total internal volume and press Label "B". The calculator then computes and prints out RT_{60} for each of the six frequencies using the Sabine equation.

Note that the highest frequency is printed first. In my test sample program, I used the coefficients for a single material as shown on the register print outs.

GENRAD 2512 PORTABLE DIGITAL SPECTRUM ANALYZER

The December 1977 issue of SOUND & VIBRATION features the new GenRad 2512 portable digital spectrum analyzer. There is an intriguing write-up by William C. Huber and Robert Sikora of GenRad, Santa Clara, CA Division.

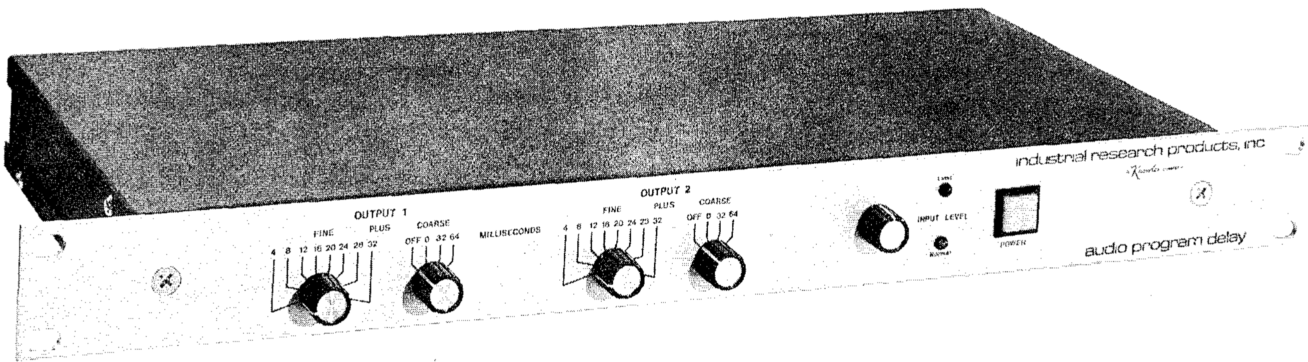
We are in contact with GenRad regarding the 2512 and have reason to believe it may finally be the FFT unit with the right combination of features to serve as a practical field unit.

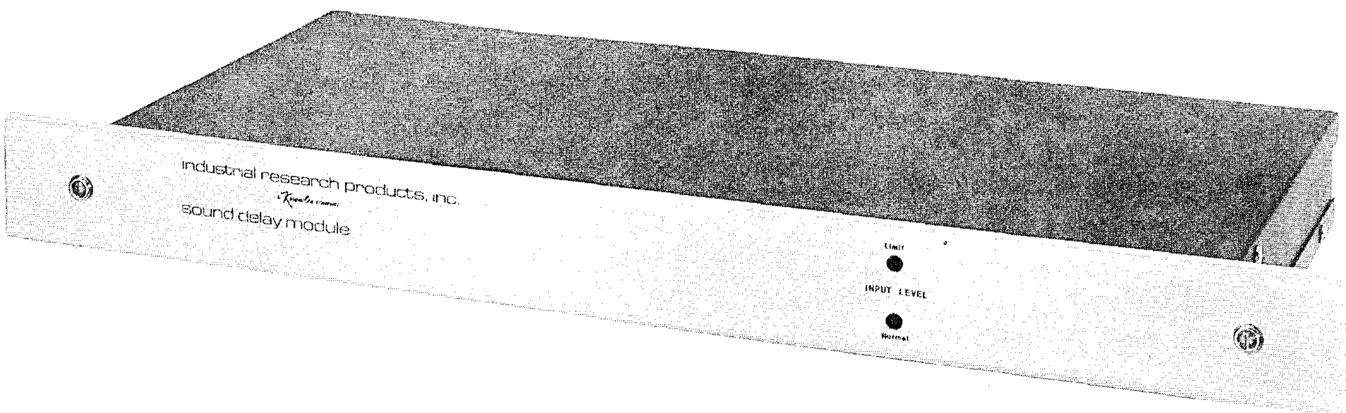
As soon as we have had a chance to test one of these units we'll report on it in detail. One feature already clearly understood and appreciated is that we are told the 2512 will sell for around $10,000.

TWO NEW DIGITAL DELAY DEVICES FROM INDUSTRIAL RESEARCH

Industrial Research Products Inc., of Elk Grove Village, Ill. has two new digital delay devices of true use and interest to installers of auditorium systems.

The DC-4011 audio program delay and the DD-4012 sound delay module offer the sound system designer a chance to both reduce system costs while raising system performance.





Using the latest advances in CCD technology Industrial Research Products Inc. has again demonstrated our contention that Mahlon Burkhard and his engineering group are uniquely equipped to assist Syn-Aud-Con graduates in staying ahead in this dynamic technical field.

If you are not now in contact with Industrial Research Prodcuts we hope these new units will encourage you to make that contact now: Industrial Research Products, 321 Bond St., Elk Grove Village, Illinois 60007. (312) 439-3600.

GARY HARRIS ARTICLES IN THEATRE CRAFTS

Gary Harris, President of G&T Harris Inc, and graduate of the last three New York classes, has had a very interesting two-part article appear in the October and November/December 1976 issues of THEATRE CRAFTS. (I know this goes back a year but most likely your local library will have copies.)

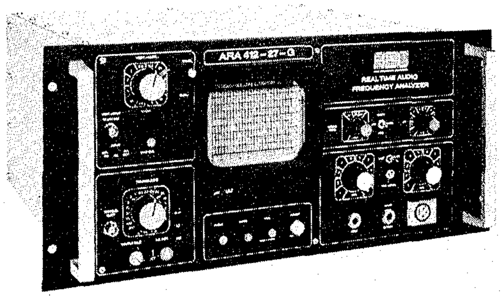
Gary has that happy combination of in-depth experience, an inbred curiosity, and solid technical background. The first of the two parts is on sound effects and other sound reproduction problems. Part II deals with sound re-inforcement challenges and their solution. The articles are a must for those handling any kind of legitimate theater.

Also, Gary holds Saturday classes on theater sound. If interested, contact Gary Harris, G&T Harris, Inc., 236 W. 55th St., New York 10019. Tel 212-581-6633.

NEW REAL TIME FROM COMMUNICATIONS CO.

Real Time Analyzer/ Oscilloscope

ARA-412-27-G



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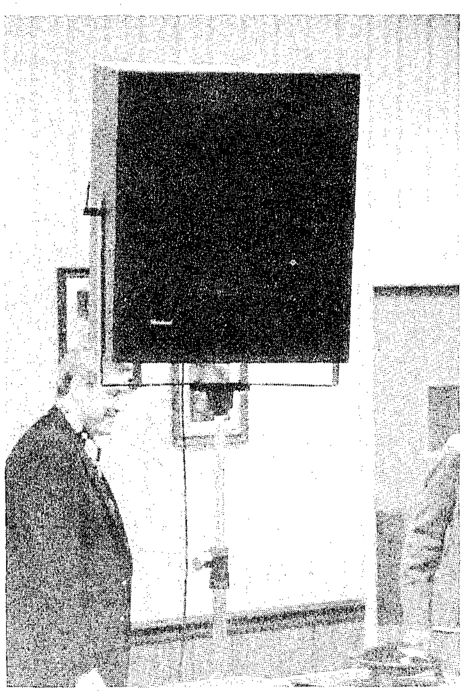
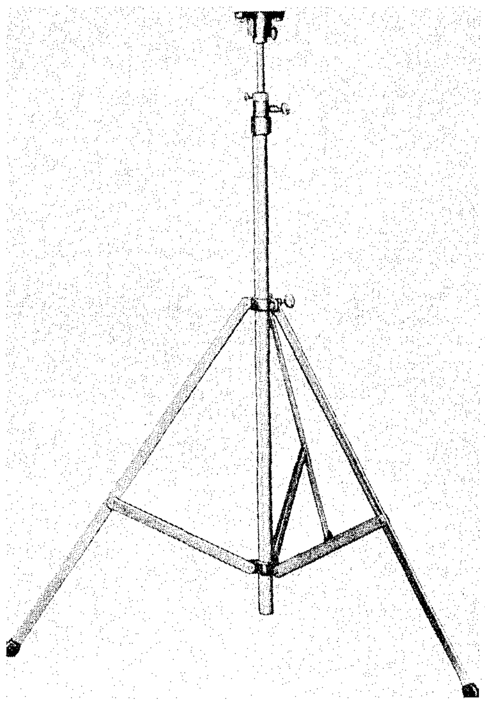
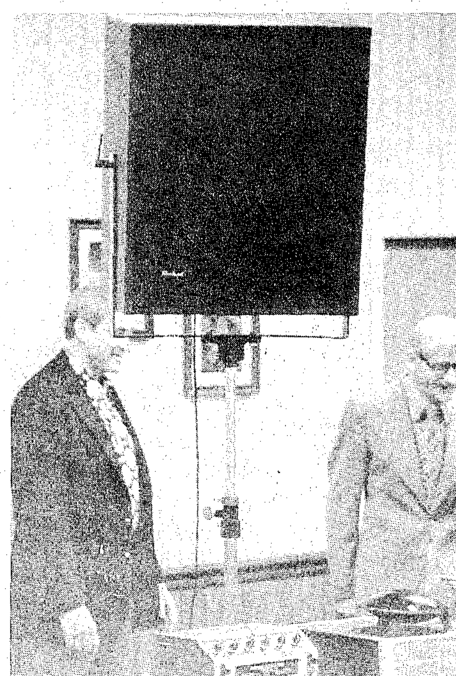
RAULAND MLS-3 SPEAKER SYSTEM AND SPEAKER STAND

Many of this year's graduates have shown an interest in the very effective Rauland MLS-3 loudspeaker system we are currently using in class plus the Atlas SS-2 speaker stand used with it.

The MLS-3 weighs 84 lbs. The SS-2 easily holds this weight securely. The units together make an excellent demonstration set up. We hope Rauland intends to market the mounting yoke they made for us to use with our SS-2.







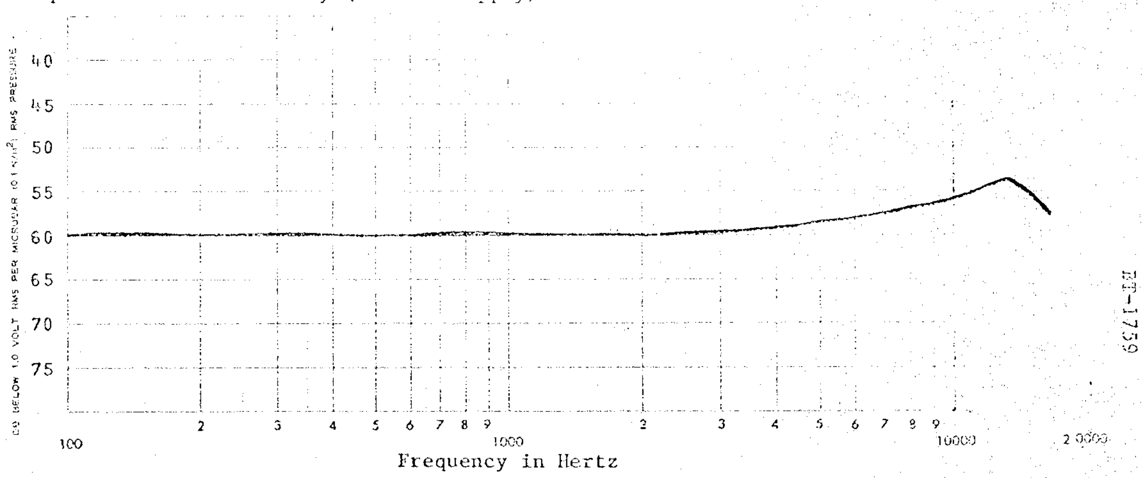
That's Don and Cec in the background enjoying a Skittle's game during a break at the Nashville class

KNOWLES SUBMINIATURE MICROPHONE

The Knowles subminiature transducer (microphone) model BT-1759 is about the size of a contact lense, is full range, and is small enough to be literally invisible when hung over an orchestra, flush mounted in a conference table, and is even small enough to find the null area (which is very small) between two out-of-polarity loudspeakers in the ceiling.

The BT-1759 has enormous possibilities. The hangup has been that the potential purchaser has to buy them in 100 lots. Syn-Aud-Con has arranged with Knowles Electronics so that we can sell our graduates sample units, one at a time, for $12 each. Use the enclosed Order Form and allow some time for the order to be filled. Knowles produces the item when ordered. If our orders in house exceed the 100 we have in-house, there could be a delay in filling your order. Each order filled will include manufacturer's literature and detailed instructions for wiring.

Open Circuit Sensitivity (1.3 VDC supply)



CLASSIFYING SOUND FIELDS

Free Fields

A sound field is said to be a free field if it is uniform, free of boundaries, and is undisturbed by other sources of sound. In practice, it is a field in which the effects of the boundaries are negligible over the region of interest. The flow of sound energy is in one direction only. Anechoic chambers and well-above-the-ground outdoors are free fields.

Diffuse (Reverberant) Fields

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.

Semi-reverberant Fields

A semi-reverberant field is one in which sound energy is both reflected and absorbed. The flow of energy is in more than one direction. Much of the energy is truly from a diffused field; however, there are components of the field that have a definable direction of propagation from the noise source. The semi-reverberant field is the one encountered in the majority of architectural acoustic environments.

Pressure Fields

A pressure field is one in which the instantaneous pressure is everywhere uniform. There is no direction of propagation. The pressure field exists primarily in cavities, commonly called couplers, where the maximum dimension of the cavity is less than 1/6 of the wavelength of the sound. Because of the ease of repeatability, this type of measurement is used by the National Bureau of Standards, NBS, when they calibrate microphones.

Microphone Calibration

Microphones can be measured in any of the above sound fields. They usually are calibrated in a free field or in an acoustic coupler (pressure field).

The type of calibration most useful in professional sound *system* work where measurements are made in typical architectural acoustic environments is the *random-incidence response.*

Microphones calibrated in a *free field* should specify the angle of incidence between the direction of sound propagation and the plane of the microphone diaphragm. Usually this is stated as 0° or 90°.

Conclusion

We suggest that for those of you with special interest in this subject and who would like to study measuring microphones in more detail that you write for General Radio's Instruction Manual, *Microphones and Accessories, VOL E.* General Radio, 300 Baker Ave, Concord, Mass. 01742

NOISE: DO WE MEASURE IT CORRECTLY?

P.V. Bruel of Bruel and Kjaer published an outstanding article in the B&K Technical Review No. 1, 1976 (reviewed in the Newsletter Vol 4 # 2), "Do We Measure Damaging Noise Correctly? We had wanted to make a Newsletter mailing of the article but were not able to obtain sufficient quantities. It has been reprinted in NOISE CONTROL ENGINEERING, March-April 1977 and the reprint is available from B&K upon request.

When we were in Texas in December, Steve Simpson III of Southwest Sound in San Antonio and a three-time grad, gave us a copy of an article by Per Bruel published in 1975, "Noise: Do We Measure it Correctly?" which seems to be a summary of the well documented "Do We Measure Damaging Noise Correctly?" It is a short article so we are reproducing the article below:

The enclosed little publication may perhaps give a first impression of being slightly confusing, as many topics are measured, discussed and analysed, but it has a historical background.

It all started because I found it curious that when we measure noise we always use the sound level meter incorrectly, since we use the A-curve also for high sound levels, and yet we obtain a more realistic result than when we use the sound level meter correctly. The explanation is not simple, but it appears that noise in octaves around 4000 Hz is more annoying than noise in other frequency regions, and since the A-curve compared to the B and C-curves accentuates 4 kHz, it is this curve of the three that gives the best correlation with our perception of annoyance. Consequently, it is quite unlikely that the A-curve should be the optimum, since many things (e.g. the D-curve) indicate that 4 kHz should be given even more emphasis.

An exact determination of how the frequency characteristics of the sound level meter should be requires extensive investigations of the human reaction to different types of noise. For aircraft noise this has only been carried out partly. This led me to examine the characteristics of the sound level meter for determination of the risk limits for hearing loss.

It has been known for more than 40 years that hearing loss always starts around 4-6 kHz (C_5-dip), and as a rule it is most severe at 6 kHz, independent of whether the damage has been caused by a single shot, firework, small explosion or similar singular events, or if the loss has occurred gradually due to long term exposure in a noisy environment. The latter is very strange, since practically all the industrial noise we know has a higher intensity in the frequency range 250-500 Hz than at 6kHz. There has been considerable speculation as to why hearing loss for industrial noise occurs 3 octaves higher on the frequency scale than the corresponding frequency region with the most energy content, and no sensible explanation has been given to date.

This led to analyses of shots from a toy pistol, signal pistol and real pistol as well as "bangs" and "clicks" from fireworks and toys. The results of the analyses of these sounds showed that the maximum energy content was in the frequency region 4-6 kHz. Further, it was established that the signal almost always consisted of one or more whole periods when equal positive and negative pressure peaks, in contrast to what was previously assumed - and described in literature - that an impulse was a short, high-level, positive pressure peak followed by a long, negative pressure pulse with much lower amplitude. It is therefore quite natural that the ear is most affected and damaged around 4 kHz by the response of a gunshot, an impulse and a click. This naturally leads to the investigation of industrial noise whether it contains the short, high sound impulses with significant energy content in the frequency region 4-5 kHz, but short enough so that they are neither registered as loud sounds by our hearing mechanism nor give a significant reading on our sound level meter.

A closer examination of peaks in industrial noise arising from hammer blows, punching, and impact between hard materials, disclosed significantly high-level but short sound impulses with both positive and negative pressure amplitudes - usually sequences of some milliseconds with large energy content around 4 kHz and with maximum peaks 15-25 dB higher than the mean value that was measured with a normal sound level meter.

Since we do not perceive the loudness of the short sound impulses even remotely correctly, one could ask: How can the ear be damaged by these impulses which sound so weak? By examining the transmission characteristics of the different parts of the ear it can be seen that the short impulses are transmitted without obstruction through both the outer and the middle ear by the nerves in the Corti organ, where the nerve ends are exposed to the full amplitude also of the short sound impulses; it is first the summing up in the brain of the sound impression that perceives a short impulse as less loud than a longer one.

It is further shown that there is a resonance amplification of 3-10 dB in the outer and middle ear of frequencies around 4 kHz. It is therefore quite natural that the damaging effects of also the industrial noise on our hearing faculty starts in the frequency region around 4 kHz, partly because by far the majority of the high noise levels are to be found in this frequency region (although we cannot hear them with their proper loudness) and partly because of the resonance of the ear at 4 kHz which further amplifies periodic sound pressures with a frequency of 4 KHz. (ED: Remember the little demo we have in each class where we resonate the ear at 3500-4000 Hz and how uncomfortable it is even though our SLM does not show a high reading. Per Bruel has explained it perfectly.)

The consequences of our finding this reasonable explanation of the mystery of the hearing threshold shift at 4 kHz is that in evaluating the damaging effects of noise and thereby setting the limits for maximum permissible noise levels, we must not only determine the sound level with a normal sound level meter, but must furthermore determine the impulsive content of the noise with a sound level meter that can be charged up very quickly.

MORE ON HEARING DAMAGE

STUDIO SOUND, September 1977 has a short artice, "Hey Man, Turn it Down" which is worthy of reproduction: More and more studio engineers are taking an interest in the possibility that hearing can be damaged by prolonged exposure to excessively loud sounds - for instance, years of monitoring at high levels. After all, who wants to be a deaf ex-engineer?

From Manchester in England and Pennsylvania State University in the USA come two interesting items of news in this field. According to Dr. T. A. Henry of the Simon Engineering Laboratories at Manchester, it may well be peaks and transients that cause damage, rather than constant high average levels. This would certainly tie in with the well-

MORE ON HEARING DAMAGE, CONTINUED

known fact that the sound of gunfire at close quarters plays both temporary and permanent havoc with hearing. And, if correct, it certainly does not bode well for studio engineers who often listen to markedly transient material (for instance, drum tracks) at high level. A spokesman for Simon Engineering Laboratories has been quoted as suggesting that the amplification equipment used in concerts and discos is relatively safe because it clips the peaks. (ED: Per Bruel makes a similar statement: "In the case of beat music, this is due exclusively to the use of electronic amplificiation which has a limited capacity to deal with high peak values. The peaks are simply cut off in the amplifier system.")

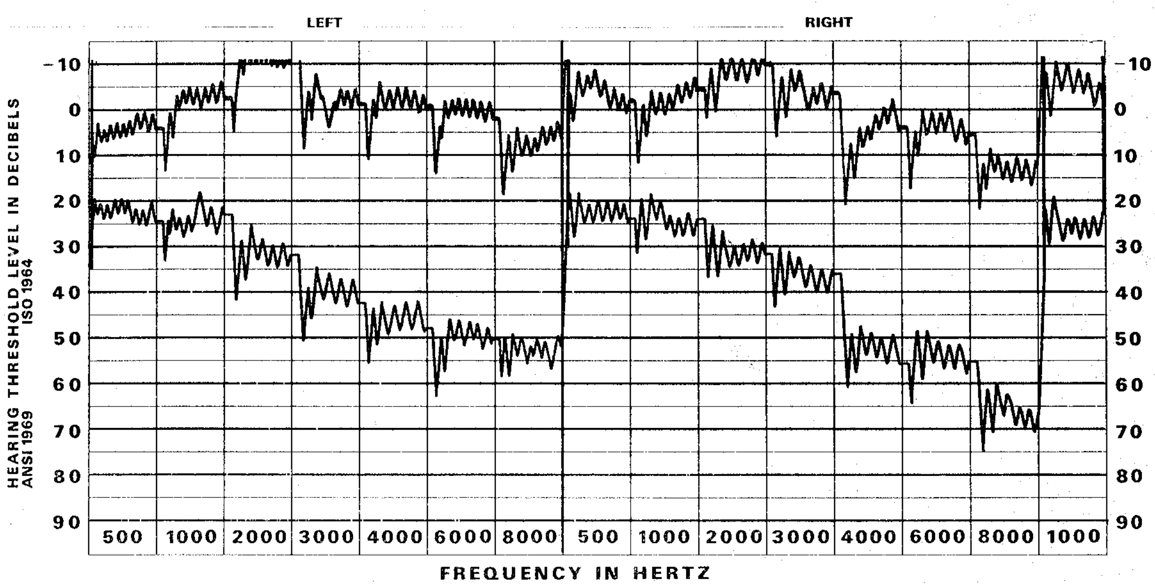
In fact this flies in the face of another equally persuasive train of thought; namely that sub-standard or over-driven equipment which clips the peaks is even more dangerous to the ears than high-quality gear with plenty of head room. The train of thought here is that when audio peaks are clipped, the waveform squares, and spikey harmonics - some of ultrasonic frequency - are produced. These spikes then barrage the ears as high-powered, high frequency transients. This theory would explain why musicians working in loud, but well-equipped rock groups seem to suffer less hearing damage than those squeezing the last dB out of budget equipment.

From Penn State University in the USA comes the suggestion that the ear is not *directly* damaged by loud noise, be it high average level, faithfully reproduced transients, or the spikey harmonics from clipping. The theory is that damage results from the human body's reaction to stress caused by noise. Apparently Penn State workers have gathered evidence that shows how under normal circumstances, when the body is subjected to loud sound, it protects the ears by releasing hormones into the bloodstream. These restrict the supply of blood to the ears and produce the kind of temporary hearing loss that anyone experiences when they listen to loud sound. Every engineer knows what sounds loud at the beginning of a session sounds far more moderate by the end.

Under normal circumstances, when the loud sound disappears, the body hormone level and hearing return to normal. But if the stress pattern is repeated too often for too long, the blood-starved cells die and temporary hearing loss starts to become permanent. The Penn State suggestion, which will appeal to studio engineers, is that, if the theory is correct, unnatural changes in the blood hormonal balance may be a signpost to risk of permanent hearing loss.

AN INTERESTING HEARING TEST

BILL PETERSON in the D.C. class tried an experiment. He wore the E.A.R. protectors home one evening after class, wore them through the night, and came into class the next day with them still in place. He took the hearing test still wearing the E.A.R. protectors. Then took the test again without the E.A.R. protectors. Bill wanted to see how much protection and attenuation he would get from the E.A.R. protectors. We had a chance to duplicate his test recently (reproduced below). The upper curve represents hearing early in the morning eithout E.A.R. protectors and the lower trace is with E.A.R.



Bill's test had another point to make: How much better would his hearing test be than a previous test taken late in the day? We know that hearing temporarily deteriorates with fatique. We had a chance to duplicate Bill's test during the DC class. The hotel notified us that our guard had walked off the job about 1:30 a.m. and while we were waiting for his replacement we ran hearing tests. The curve was down about 5 to 10 dB from early morning runs.

BILL PUTT from New York brought into the Orlando class a study prepared for the Department of Transportation and we are reproducing three of the tests. Note that cotton is better than nothing to protect your ears and they make the point that wet cotton is even more effective than dry cotton, which was used for the test.

(see next page for tests)

HEARING TESTS, CONTINUED

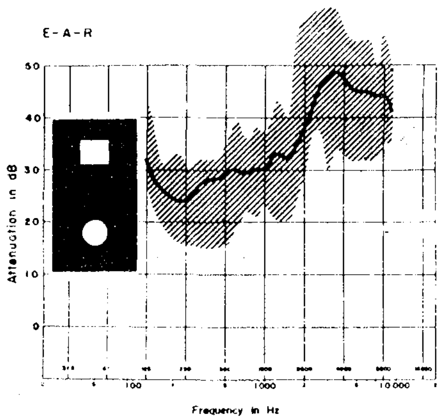


FIGURE 1. Measured attenuation provided by E-A-R. The line represents the mean attenuation; the shaded area represents the range

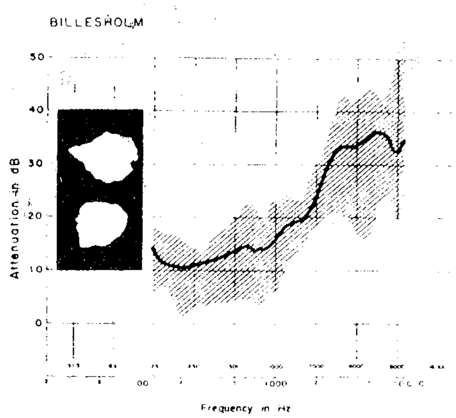


FIGURE 11. Measured attenuation provided by Billesholm Swedish Wool. The line represents the mean attenuation; the shaded area represents the range

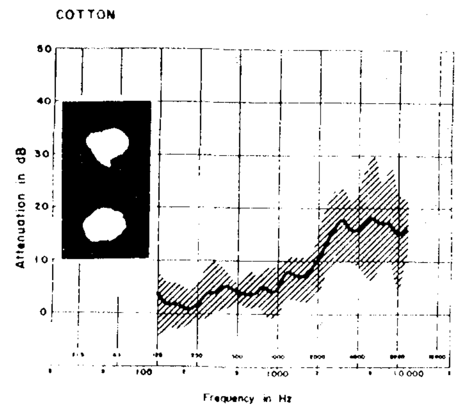


FIGURE 21. Measured attenuation provided by Johnson & Johnson cotton wool. The line represents the mean attenuation; the shaded area represents the range

DERIVING THE SABINE REVERBERATION EQUATION

First imagine a totally enclosed room that you are going to fill with sound energy. The larger the room, the greater the volume, V, the more sound energy one can pour into it before overflow. Therefore, if one were to let energy flow out of the room through a hole the larger the room the longer the time it would take to empty the room of energy through a given opening. Thus:

$$T :: V$$

Where :: means proportional to

It is again relatively intuitive that the larger you make the opening, the great its absorption of the energy would be (it could pour out through the opening faster), therefore:

$$T :: \frac{1}{\bar{a}}$$

Where $\bar{a}$ is the average absorption coefficient present

As the sound energy is reflected about, it *tends* toward an average distance between encounters with boundary surfaces. This distance is called the Mean Free Path and has been found both mathematically and emperically to be:

$$MFP = \frac{4V}{S}$$

If sound travels approximately 1130 ft/sec., then the MFP divided into the velocity of sound would give us the number of reflections per second (RPS)

$$RPS = \frac{MFP}{\text{Velocity of sound}}$$

The number of reflections (N) that are normally taken into consideration in calculating or measuring reverberation time are those reflections that occur while the sound level is dropping 60 dB after the sound source is turned off

$$\frac{N}{RPS} = RT_{60}$$

Where RT_{60} is the reverberation time in seconds for the sound level to decay 60 dB

Now, 60 dB is $\frac{1}{1{,}000{,}000}$th of the original energy

$$e^{(6 \ln 10)} = 1{,}000{,}000$$

Therefore, we could write N as

$$N = 6 \ln 10 \frac{1}{\bar{a}}$$

If we now write our equation the long way, we find that

$$RT_{60} = \frac{N}{RPS} = \left[(6 \ln 10)\frac{1}{\bar{a}}\right]\left[\frac{1}{1130}\frac{4V}{S}\right] = \frac{.049^{*}V}{S\bar{a}} \qquad {}^{*}.049 = \frac{4(6 \ln 10)}{1130}$$

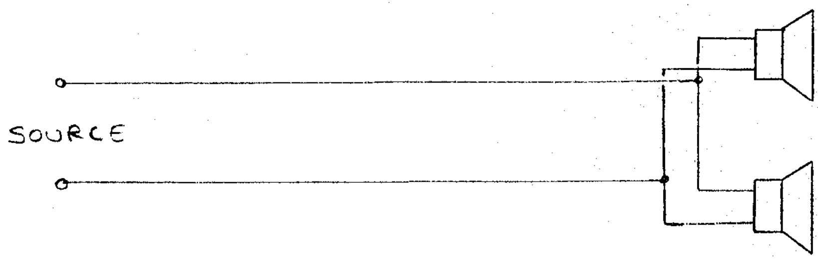
which is the classic Sabine equation.

DISTRIBUTED LOADS

BY

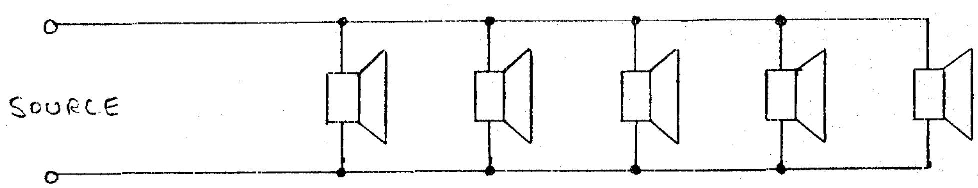
ED LETHERT

As we go about the business of installing loudspeaker systems, we generally find them connected in one or two configurations. The first and simplest would be the concentrated load, typically:



Here the load is concentrated at the far end of the line and line losses would be calculated in a normal manner, all speakers being considered together as a single load.

The second configuration, which is always found in distributed sound systems, is the distributed load. Typically,



Now, we find the load distributed along the line, and intuition would tell us that losses should be somewhat less than if the same total load were connected at the far end of the line. The question has always been, is there a simple way to determine the actual line loss of the farthest speaker? While such calculations might not be necessary in many cases where loads are small and lines relatively short, let's take a real life example where the difference is considerable.

Let's assign some values to our distributed system shown above. Assume that the speakers are set at 32 watts each. The distance from the source to the first speaker is 400 ft, and the distance between successive units 400 ft. Our total line length is therefore 2000 ft. The total load is 160 watts.

Up to now we had to move all loads to the end of the line to calculate line loss. In doing so, we would find that if we used #12 wire, our line loss would be 1.6 dB. In order to reduce it to 0.5 dB or less we would have to use #6 wire. Obviously, copper costs will be high, and larger raceways will be required.

Now we can apply a method which has been used by electrical engineers for years and it involves a factor called the LOAD CENTER. The load center is actually a point located along the line which would be the equivalent line length if the entire load were concentrated there.

The procedure for calculating the load center is quite simple

1. Multiply each load (watts) by its distance from the source (in feet). Do this for each unit.
2. Add up all of the above products.
3. Divide the sum of the products by the sum of the individual loads (total watts).
4. This gives you the load center. The line loss is then computed as if the system were a concentrated load located at the load center.

Let's apply the values from our distributed system and see how it works.

Example

32 watts x 400 ft = 12,800
32 watts x 800 ft = 25,600
32 watts x 1200' = 38,400
32 watts x 1600' = 51,200
32 watts x 2000' = 64,000
192,000

32 x 5 = 160 watts

$$\frac{192{,}000}{160} = 1200'$$

The load center is therefore 1200 ft. We can now make our calculations assuming a 160 watt load concentrated at 1200 ft.

We find our line loss for #12 wire is 1.0 dB, for #10 it is 0.6 dB and for #8 it is 0.4 dB. It is worth noting that the calculated loss applies to the farthest load, and that the loss decreases with each load as we move back towards the source.

I think it has become obvious that the few calculations we made here can result in tremendous savings, and at the same time they provide us with an accurate indication of our system's line losses.

NASHVILLE AND ORLANDO - 1977

A WASTE OF TIME

I usually consider it a waste of time to write letters to Hi Fi magazines regarding flagrant errors as the editors sometimes do not have someone to turn to that knows a fact from a flower. Occasionally a Hi Fi editor will check technical facts by committee vote and whoever receives the most votes wins. Hence errors can creep into the sacred pages of the publication. (We all have that problem.) Often "respected equipment reviewers" are not as well informed about all aspects of the equipment they are reviewing, which is understandable as they take on a myriad array of equipment. Therefore it behooves the reviewer to be in a learning mode at all times and to be receptive to new input. I know that we look forward to every class with eagerness as we know that we are going to learn something new; there will be several people in that class that know more about some given subject than we do. That was the point that I was trying to make to Mr. Eisenberg in my second letter. cd

August 2, 1977

Letter to the Editor
Modern Recording
14 Vanderventer Ave
Port Washington, NY 11050

Sir:

I read Len Feldman's AMBIENT NOISE "BiAmping, Tri-Amping and Such" in the August 1977 issue of MODERN RECORDING and I thought, "hey, that's wonderful. The Hi-Fi world has got the message straight on bi-amping." I turn the page and read Norm Eisenberg and Len Feldman's Lab Report on equalizers. The Hi Fi boys need an education on that subject now. Writing on the Delta-Graph EQ-10, "The results are impressive in that they show the lack of interaction of adjacent controls, a characteristic not always managed by less sophisticated equalizers in which opposite settings of adjacent controls tend to neutralize or cancel each other out. What this indicates, simply, is that the EQ-10 can provide a fine degree of control to permit tailoring overall response to specific needs of program material and/or acoustic environment."

And, writing on the Spectro-Acoustics Model 210, "Finally, to illustrate the precision with which the model 210 can create a given response curve, we photographed (Fig. 4) the actual response curve obtained when the ten controls are adjusted as shown in Fig. 1."

I would like to ask Mr. E and Mr. F one question, "Why do all the expensive and sophisticated equalizers interact (Altec, UREI, etc.)? When they have the answer to that question they will know a lot more about equalization than they do now.

I don't really mind that they don't know much about it now but such reviews encourage manufacturers, who often aren't applications oriented, to continue to go through the costly process of designing and building a product that doesn't properly meet the needs of the field.

Sincerely,

Carolyn Davis

(The following is Norman Eisenberg's response as published in the December issue of MODERN RECORDING.)

Apparently, pet concepts about graphic equalization are becoming as fashionable as pet notions in other audio areas. But what works for one audio person in a given acoustic situation may not apply to all others, and so there always is the danger of elevating a pragmatic solution to a particular problem to the dignified status of a "scientific truth." Be that as it may, it would seem self-evident that one of the criteria of good circuit design in which several frequency-critical segments are strung together is the capability of that circuit to produce its intended results as a complete device, and also in terms of the scientific action of each of its integral segments. So, the extent to which segment A degrades the action of segment B could be taken as a "limitation" on the whole.

If such a device satisfies some audio need, fine. But other devices - designed to a somewhat alternate philosophy - should not be penalized as a result. As for "sophisticated equalizers," the UREI is unfamiliar to us, but we have worked with and tested many others, including the Altec which has been around for at least eight years. In MR's tests of recent equalizers we have shown response characteristics that looked fairly like those reported on the Delta-Graph. If we are to believe the writer of this letter, then in addition to Delta-Graph, such companies as Crown, bi-amp, Soundcraftsmen, Klark-Teknik, et al also do not understand equalization. This seems to us a rather untenable and unprovable position.

Norman Eisenberg
Audio Editorial Board
Modern Recording

November 15, 1977

Letter to the Editor
Modern Recording
Port Washington, NY

Dear Sir:

I regret that Mr. Eisenberg responded so vociferously to what I felt was a gentle reminder that he needed to do further research into the "workings" of filter sets. (December issue)

(Continued next page)

Letter to the Editor, continued

Mr. Eisenberg errs in labeling filter interaction as a "fashionable pet notion" and then further pontificating that "there is always a danger of elevating a pragmatic solution to a particular problem to the dignified status of a scientific truth." The Ancients, when proving the earth was flat, used Mr. Eisenberg's "It would seem self-evident" as an opening ploy.

The very point of my remarks in my earlier letter is that what he calls a "rather untenable and unprovable position" is merely the manifestation of his lack of experience in what equalization is all about and that a closer look at the fundamentals would do him no harm.

In spite of his overwrought reply I would, for the sake of your readers, like to answer the question I asked him to consider, "Why do all the expensive and sophisticated equalizers (Altec, UREI (which Mr. Eisenberg never heard of), etc., interact?"

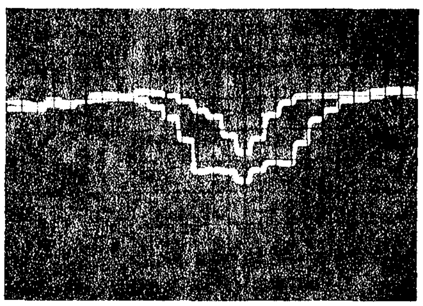
The first misunderstanding a competent circuit engineer sometimes makes in designing an equalizer for use in adjusting electroacoustic transducers to acoustic environments, be it for home use or professional use, is to assume that

1. There is a "real" signal that can be boosted or attenuated
2. The bandwidth may be arbitrarily determined, as well.

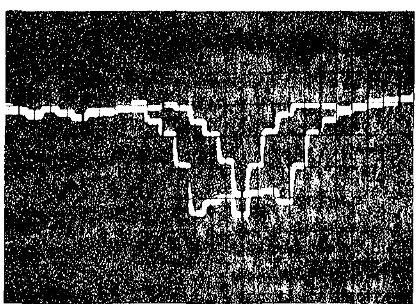
Both of these assumptions ignore the facts that, first, no two signals can come together in an acoustic environment to an addition greater than +3 dB but that they can come together to attenuate to any depth; and secondly, that the very narrow band effects observed in playback systems are usually non-linear resonance effects not amenable to equalization.

Once the neophyte to acoustics, albeit seasoned circuit designer, finds out that he can't boost diaphramatic absorptions, phase cancellations, and coincidence dips and that the first step in any successful equalization is to replace the devices that have excessive non-linear resonances, then the beginning of an understanding about filters leads him away from parametric non-interactive devices into a new look at the professional interactive devices. (This is not to say that parametrics do not have a definite use for special effects and they can be particularly valuable in rental-temporary sound systems.)

An interactive, or combining filter, is one in which any two adjacent filters interact so that if one filter is set at -4 dB at 1,000 Hz, a second filter set at -4 dB at 1,250 Hz, the result is not two dips but one smoothly shaped dip having a total attenuation of -8 dB and a center frequency *exactly between* the two labeled frequencies. This is an important test that should be part of the curves you show in your tests.



Combining filters



Non-combining filters

The combining or interactive filter is called a "minimum phase filter" and allows the following very real benefits.

1. The attenuation can appear anywhere inbetween the label frequencies of any two filters.
2. Smoothly shaped curves can be obtained that are the inverse of the acoustic amplitude response.
3. When the amplitude differences are brought to zero between all frequencies *so are the phase differences.*

In the same issue of MODERN RECORDING in which Mr. Eisenberg answers me with "..it would seem self-evident that one of the criteria of good circuit design in which several frequency-critical segments are strung together is the capability of that circuit to produce its intended results as a complete device, and also in terms of the scientific action of each of its integral segments, so the extent of which segment A degrades the action of segment B could be taken as a 'limitation' on the whole," he writes on page 23 of the White Passive Equalizer model 4004, which certainly can be added to my list of professional equalizers along with the Altec and UREI, "responses of any two adjacent sections *add smoothly* (italics mine) without response curve 'ripple'." Mr. Eisenberg calls the White equalizer "designed for professional sound reinforcement applications", and that is exactly the point I am making, two adjacent filters on a professional equalizer should interact, combine, or, if Mr. Eisenberg prefers, "add smoothly".

Mr. Eisenberg is contradicting himself in the same issue of MODERN RECORDING, which indicates a need to know more about filters and their characteristics before reviewing for a professional audience. And I am making the assumption that MODERN RECORDING's readership is made up of people in professional audio.

Fundamental filter theory goes back to McElroy (1934), Kimball (1937) and Terman (1943). Mr. Eisenberg should familiarize himself with some of the basic literature available on this subject. I would recommend *RADIO ENGINEERS HANDBOOK* by Frederick E. Terman, McGraw Hill, 1943, pages 218-219 on "The Phase Area Theorem" and the chapter on Circuit Theory. Also, *SOUND SYSTEM ENGINEERING* by Don and Carolyn Davis, Howard W. Sams, Inc., Third Printing 1977, pages 142-151, and the writings of Dick Heyser in AUDIO Magazine and the JOURNAL of the AES.

One further test that you should make on filters, the one you allude to in your write up of the White 4004 when you say "two adjacent sections should add smoothly *without response curve 'ripple.'"* (Italics mine). Most filters are not tuned to the correct center frequencies. Most are tuned to the ISO labels whereas the labels are intended for just that, "labels", and the filter so labeled should have been tuned to the *exact* Renard number center frequency (i.e., an 800 Hz 1/3-octave filter should be labeled 800 Hz but tuned to 794.33 Hz, etc.)

I hope that you do not take my comments personally but can benefit from them for the good of yourself and your readers.

Carolyn Davis

(continued next page)

Letter the Editor of MODERN RECORDING

December 7, 1977

Ms Carolyn Davis
Synergetic Audio Concepts
P O Box 1134
Tustin, CA 92680

Dear Ms. Davis,

We are in receipt of your letter dated November 25, 1977 concerning equalization filters. Copies of your letter have been forwarded to both Mr. Eisenberg and Mr. Feldman. While we feel that many of your remarks are valid, we sense that a continuation of this discussion in print would no longer be beneficial to our readers. Therefore, your most recent correspondence and any reply possibly forthcoming from Mr. Eisenberg will not be printed in Modern Recording.

We feel that this is advisable since the letters are getting to the point of personal attack.

We can assure you that Mr. Eisenberg has heard of Altec and UREI equalizers as well as a good many others, as he is considered to be one of the most respected equipment reviewers in the country.

Mind you, this doesn't mean that any future correspondence from you on other topics won't be printed. We have welcomed your remarks in the past and have the utmost respect for the work of Don and Carolyn Davis. This policy applies to this instance only.

Thank you for your interest in Modern Recording.

Sincerely,

Modern Recording

Hector G. La Torre
Editor

cc: Norman Eisenberg
Leonard Feldman

MORE ON AZIMUTH ALIGNMENT OF TAPE RECORDERS

Bill Reventos, many years with Electro-Voice and now Product Director at Ivie Electronics, writes:

I was very interested in the article by Andy De Ganahl from Disney World. We've been playing with aximuth alignments and IE-10As for some time here, and the added insight coming from Andy's obvious practical experience in the matter gave us interesting new information. (Newsletter Vol 5 # 1, October 1977)

Incidentally, here's a tip we've learned for azimuth alignments of the record heads of stereo and multitrack machines. Unless one has an oscilloscope handy the alignment is usually done by looking at some sort of meters indicating the output of left and right channels, the idea being to adjust azimuth until output from both channels in the highest frequencies is maximized without favoring one channel or the other.

Using one IE-10A it is possible to accomplish the same thing by combining the outputs of the two channels in question ELECTRICALLY OUT OF PHASE into the input of the IE-10A. Now, after injecting pink noise into both inputs, the aximuth is adjusted by seeking the greatest NULL in the 8 and 16 KHz octave bands. This method works very quickly and is surprisingly accurate on the tape machines that we have tried. Assuming one can obtain an accurate tape of pink noise, it is possible to first align the play back heads by the same method.

Multitrack machines would simply take two nonadjacent tracks (not outside tracks) for the same procedure. While the approach may be somewhat naive, it appears to work very well.

There are several alignment tape manufacturers who offer pink noise alignment tapes including Standard Tape Laboratories Inc. and Magnetic Reference Labs, both in the San Francisco Bay area, I believe. Of the two, I can probably more highly recommend Magnetic Reference Labs. Getting into the idea of pink noise tapes in general leads to some interesting differences between tape recorders. Any of the Standard Tape Labs or Magnetic Reference Labs people can give you an interesting and entertaining discussion as to why a "flat" pink noise tape only works perfectly on the type of tape machine it was recorded on, but I'm afraid I'm not sufficiently versed in the facts to give you an outline in this letter.

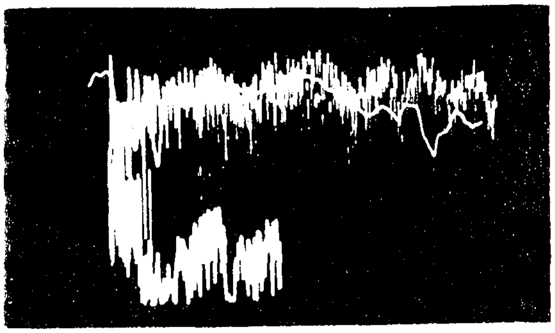
The best and most "foolproof" way to obtain a pink noise tape for your particular machine is to go through all the trouble of aligning it very meticulously with a standard alignment tape *once*. Then, when the machine is as perfectly aligned as you can do it, make your own pink noise tape with a good "flat" pink noise generator like the IE-20B.

ANOTHER ANSWER FOR NYYA LARK'S QUESTIONS

Bill Reventos also wrote: I'd like to pass along a comment about the article on drummer isolation (Vol 5, # 1 Newsletter). One of Nyya Lark's questions had to do with a television orchestra conductor using a headset. Many times the TV orchestra direction uses a headset not to hear a particular portion of the mix of his orchestra, but rather to receive direction as to when and what to play. This is evident in many of the talk shows and other shows using large live orchestras. Also, it is fairly commonplace for the conductor to wear a single cup headset (the drummer may also wear one for the same reason) to listen to a "click track" when the live orchestra is being "synced" with something on tape. The traveling Disney Spectracular Shows of a couple of years ago were examples of the latter; they had a relatively small band on the road with them, which was augmented by taped brass, strings and special effects. The conductor would wear a headphone to monitor the click track so he could keep the live musicians synced with the tape. The effect is of having a large live orchestra in the house!

TDS AND CRITICAL DISTANCE

An interesting example of the combination of loudspeaker Q and acoustic absorption both increasing in value as the frequency increases is shown in the spectrum analysis shown below.



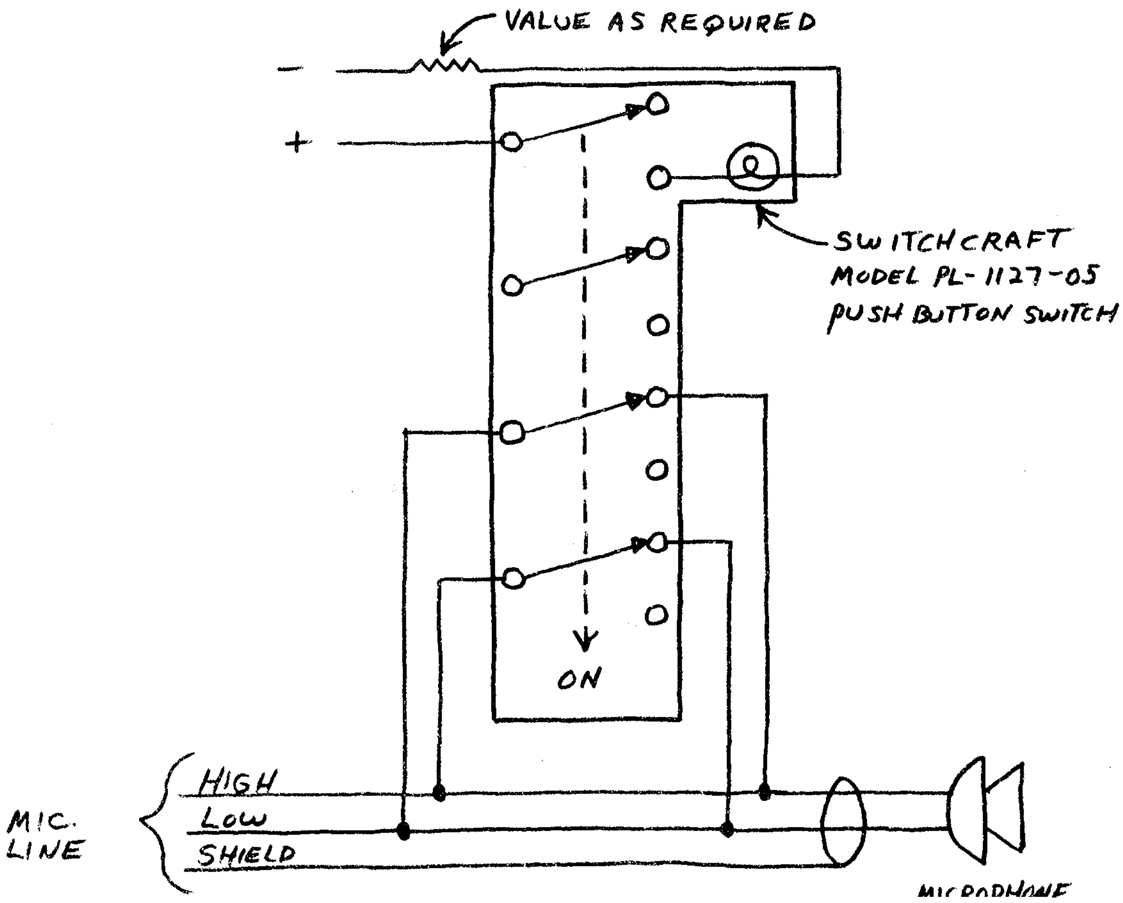
The jagged spectrum is the *total sound* entitled by the loudspeaker as measured near what we felt D_C should be in this environment.

The solid line spectrum running through the middle of the jagged plot at low frequencies and dropping below it at high frequencies is the *direct sound* from the loudspeaker. At the high frequencies the measuring position *is* beyond D_C while at lower frequencies the measuring position is either at or inside D_C.

The bottom jagged partial trace is not part of the measurement but a residue of the calibration process.

QUIET MICROPHONE SWITCHING STANDARD

ROBERT REIM, of Acromedia, Culver City, CA, is an experienced audio engineer who has been involved in the solution of many, many basic but plaguing problems. Recently a Syn-Aud-Con grad asked us for the answer to *quietly* switching microphone circuits, especially in conference systems. We referred him to Bob who returned this answer.



ACROMEDIA DRAWING # A-693

USEFUL VARIATION OF THE EPR EQUATION

A very useful variation of the EPR equation found on page 86, eq. 5-43 of *Sound System Engineering* is one BILL SYMMES of Bogen rearranged so that he could find out the minimum sensitivity he was looking for in a loudspeaker if the electrical power available was restricted:

$$\text{Min. } L_{sensi} = (SPL_D + 10) + (\Delta D_2 - \Delta 4') - 10 \log (EPR)$$

Where $SPL_D + 10$ is the sound pressure level desired plus 10 dB of meter lag margin

EPR is the electrical power required that is available

ΔD_2 is the D_2 distance (loudspeaker to listener) converted into dB)

AMPS, AMPS EVERYWHERE BUT NARY A WATT - I THINK!

BY

BILL PETERSON
PROFESSIONAL SOUND
FALLS CHURCH, VA

The Wednesday of the job was one of those "made for Murphy" sort of days. To begin with, my 240Z ran into a guard-rail, neatly compacting the front end, forcing me with record player and analyzer in hand to hop a bus for fifteen blocks. Returning with a borrowed vehicle to get the rest of the equipment from my incapacitated car, I discovered that it was missing. Returning to the job site, I found everyone avidly discussing the placement of twelve sections of scaffolding with no one making any progress toward actually getting it erected. Musing over my sprained hand, acquired when the engine had attempted to join me in the drivers seat, I stood there considering the left-behind stage box for the snake, and all the amplifier output adapters keeping it company in the shop.

Ever been there? Sure you have! When you're right at the point of saying chuck it, but you know that you can't, but you want to anyway. You call upon that reserve we all have, and jump back in there with both feet - bound and determined to see that it all comes out right. In this case, however it never really did, and the lesson I learned took four months to sink in.

But, back to my story. We got the scaffolding in place and a whole big pile of speakers on top thereof, and the snake stage-end arrived from the shop along with the amplifier connectors. The truck was turned around facing away from the stage area with its open rear door, and the 5 kw generator was placed just inside.

With the nearly uncountable delays, we were hooking up the last of our equipment when the orchestra started arriving. Along with them came sixty stand lights of 25 watts each, but fortunately someone had made provision for that with an additional 4 kw generator. This we placed alongside the one we had for sound in the back of the truck. They both started right up and purred along - about the first thing that had gone right the whole day.

With throngs of people now beginning to mill about, we gave thought to how nice a cordoned off area for our equipment would have been, but that also was out at this late time.

Trying to equalize the system, we discovered that whenever we slipped into feedback the lights on the mixing console dimmed. In addition, the power amplifiers, of which we had quite a few, gave forth with a most pronounced groaning sound as their indicator lights also winked off.

Thinking there might be some problem with the generator, Bruce went back to check and baby-sit them. With still no explanation as to what was happening, the orchestra leader picked up his microphone and the concert began.

Three-fourths of the way through the National Anthem, the stand lights blinked twice, then a third time, then went out completely. I was at the console, and assumed someone had unplugged their extension cable to the generator. Figuring Bruce would have this fixed in a few moments, I waited patiently along with 10,000 people and a rather nervous orchestra. Two minutes went by, and I proceeded to elbow my way through the multitudes toward our truck. There, perched on the top of the cab was Bruce, calmly awaiting the righting of whatever it was that had interrupted the program. Yelling to Bruce, I ran to the rear of the truck only to find the roll-up door rolled down. Jumping up on the truck, I yanked open the door to a box of air totally devoid of oxygen. The orchestra light generator had starved itself before our unit gave up, and dragging it to the edge of the opened door, with a couple of pulls, it churned back to life. I must have looked at Bruce rather strangely, for without my asking he proffered that the noise had bothered him, so he had simply closed the door. The list was getting longer.

The orchestra finished their performance, but we were never able to get any gain. The LEDs on the Bose 1800s barely flickered. Determined to show all present the power of this enormous system we had assembled, we plugged in the record player I had salvaged from my car and proceeded to play music to the departing audience.

Of course, this time we had no gain problem, but once the second LED on the amp came on, the lights dimmed, the generator slowed down, needless to say, the frequency dropped, and our record player turned Frank Sinatra into Larry Hooper.

With this final effort at saving at least some face, now fully aborted, I could only look forward to a warm bed still some four arduous hours away.

At a meeting of the clan the next morning, we discussed the prior evening's happenings, and Bruce apologized for his oversight with the generator, and the rest of us took turns apologizing for our own monumental goofs. No one, however, had the slightest idea what had been the problem with the power, although we all recognized the pure idiocy of the idea (it was mine) of attempting to run a frequency dependent thing like a turntable off of a generator.

It was one of those evenings you try to forget, and I did a good job for nearly four months, when driving down the road one day (in my new car) the explanation of what had been the power problem finally came to mind.

First I should tell you that we had wanted a high degree of control over our loudspeaker coverage, so had placed just a couple of speakers on each of several amplifier channels. In fact, we had used three Bose 1800s, two BGW 500Ds, and two BGW 250Bs. If you ever have the occasion, which I have, to run power efficiency curves on amplifiers of this sort, you will find them on the order of 1% efficient at a one watt power output. Therefore, in order to put out just one average watt from each of these amplifier channels, the a.c. power input requirement would be 700 watts. Even so, it would appear that our 5 kw power source would allow seven watts from each of fourteen amp channels, giving us at least a respectable power output; but think further.

Remember in live music, we are dealing with at least a 13 dB crest factor. Some might argue that it is really higher, but for the sake of mathematics I usually use 13 dB. (13dB represents an easily rememberable power ratio of 20 to 1). So, with our 13 dB crest factor, and a 5,000 watt maximum power capability available, it is easy to see that our average power was only 5,000 divided by 20, or 250 watts. This was it, period. The most we could hope for before

(continued on the next page)

AMPS, AMPS EVERYWHERE....BY BILL PETERSON (CONT)

our generator started loading down on transients. Remembering that in order to produce one electrical watt from each amp channel, we needed 700 watts of a.c. power, you can see that we could get but a third of a watt from each amp channel, or only 5 watts total.

Quite interesting, isn't it? Only five available average watts from 5,000 watt power source, but believe me, that is the way it is, or I should say, was! In retrospect, we should have series paralleled all of our speakers on one, or at the most, two of the smallest amplifiers. We would have had far more power output capability. Also, if you should find yourself in a similar position of trying for the last bit of efficiency, it is interesting to note that an amplifier delivers its highest efficiency into a high output impedance. I would have thought just the opposite, but experimentation shows a greater efficiency at 16 ohms than at eight or four.

One other observation regarding generators: I, and you, have probably also run a whole bunch of amplifiers off of one little 20 amp breaker. In fact, I ran a French rock-musical once with seven Bose 1800s and one BGW 500D amplifier on one 20 amp circuit, and it was LOUD. And this with only 2,500 watts, but 2,500 watts RMS. If you've ever gotten a screwdriver across the dimmer you were trying to install in your bedroom, you've realized that you can draw a whole bunch of amps for a fraction of a second. Your a.c. power outlet then has its own built in crest factor margin which gives you a nice chunk of power for those peaky transients. This, the portable generator lacks. When it says 5 KW, that means 5 KW and that's all there is, there ain't no more.

Most, probably the majority of you, have not ever gotten yourselves in the position of having more amplifiers than brains, but as they are getting cheaper all the time (amps, not brains) my learning experience might just have occasion to save you from a few pitfalls.

(ED NOTE: We live 7 miles from the nearest power line - in the Santa Ana mountains. We live off a 5KW generator. There are a lot of limitations to generator living, and we know about most of them.)

IMPEDANCE MATCHING IN LINK CIRCUITS

The connection between a microphone mixer amplifier and a power amplifier, as a typical example, is termed a "link circuit". That is, it "links" one device with another device electrically.

In the most common case, the mixer is marked as a 600Ω output impedance and the power amplifier is marked as 600Ω input impedance. Most Syn-Aud-Con graduates know, or should know, that while marked 600Ω output, the mixer amplifier will, if measured, turn out to be closer to 100Ω. Many wonder why the industry follows the convention of making outputs much lower in impedance than their labels and inputs much higher in impedance than their labels.

This engineering technique is better understood when it is recognized that a mixer that was 600Ω connected to an amplifier that was 600Ω would equally divide the signal voltage between the two impedances. That is to say that if you measured 1 volt across the output of such a mixer with a very high impedance voltmeter before it was connected to anything and saw a reading of 1 volt, then upon its being connected to a matching impedance, the voltage output would drop to .5V (-6dB). This all assumes, of course, that the mixer has a constant voltage output and that the source can deliver the current called for in such circumstances.

The diagram shown provides the calculation of each of the parameters involved in such a case. Note how much higher the source voltage, E_S, must be in order to provide a load voltage, E_L, that meets the required power in the load, P_L, when the source impedance, R_S, is made equal to the load impedance, R_L.

Attempts to drop load impedance down to true source impedances usually results in greatly diminished output and a drastic rise in distortion.

Two useful rules to consider when matching devices are:

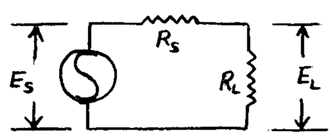
1. Make the load impedance at least ten times the source impedance

$$R_L \geqq 10\, R_S$$

2. If you can't make $R_L \geqq 10\, R_S$ then pick an R_L that does not drop the output of the source by *more than 2 dB* when R_L is connected.

Making R_L much greater than R_S results, normally, in reduced level being transferred to the next active device without the penalty of increased distortion. Making R_L approach R_S causes reduced output and high distortion.

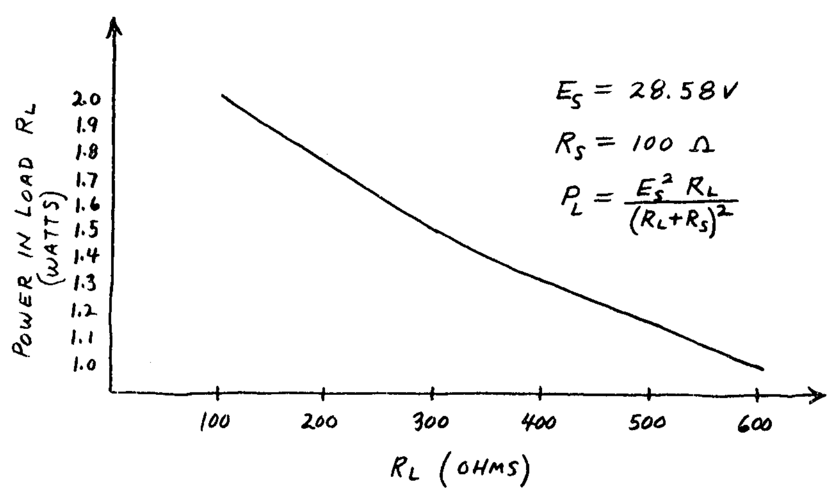
BASIC POWER TRANSFER CALCULATIONS



$$P_L = \frac{E_L^2}{R_L} = \frac{E_S^2\, R_L}{(R_L + R_S)^2}$$

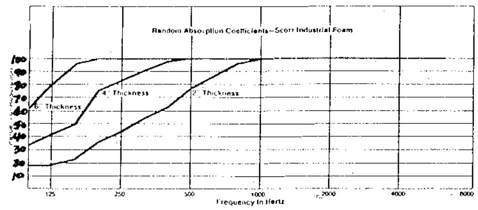
$$E_L = \sqrt{P_L\, R_L}$$

$$E_S = \sqrt{\frac{P_L (R_L + R_S)^2}{R_L}}$$

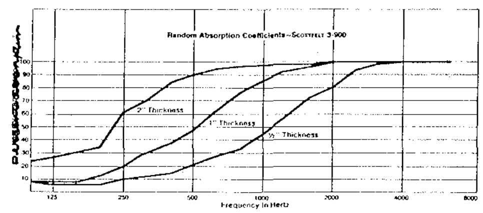


SCOTT ACOUSTICAL FOAMS

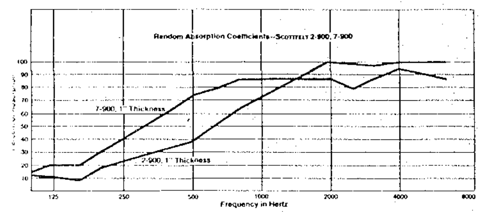
NELSON MEACHAM, of WED Enterprises in Glendale, CA and graduate of the 1977 Orlando class, sent us some very useful data on Scott Acoustical Foams



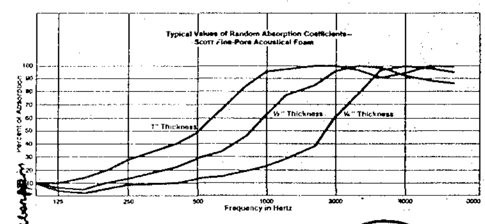
Scott Industrial Foam
The graph, left, illustrates sound-absorption capabilities of Scott Industrial Foam in three thicknesses. Because of the uniformity of fine-pore size and open-pore structure, the absorption results will remain constant and predictable from sample to sample, application to application. Note that increased thicknesses provide greater absorption in the lower frequencies, and will not decrease once 100% absorption is accomplished.



Scottfelt 3-900
Where space is critical, Scottfelt is particularly valuable. It provides unique, predictable absorption qualities in a limited thickness. The representative curves, left, show the consistently good absorption values of Scottfelt.



Scottfelt 2-900, 7-900
Shown, left, are two different grades of Scottfelt. These curves illustrate how the material can be altered to match specific requirements for absorption in a particular frequency range.



Scott Fine-Pore Acoustical Foam
Scott Fine-Pore Acoustical Foam offers a valuable combination of good acoustical properties, physical characteristics, and cost. The curves shown are typical values based on an arithmetical average since some variation will occur.

SCOTT

Foam Division, Scott Paper Company
1500 East Second Street
Chester, Pa. 19013
Telephone: 215-876-2551

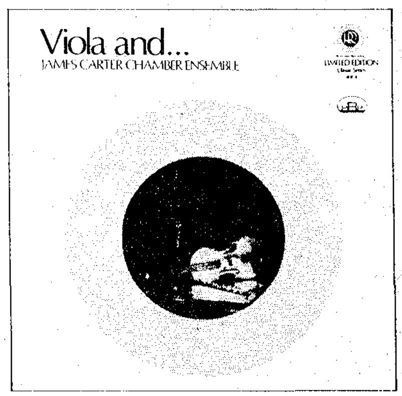
Form No. 3655

Litho in U.S.A.

These materials are seeing very wide industrial use everywhere from aircraft to microphone windscreens.

REFERENCE RECORDS

NELSON MEACHAM also included the listing for the first three Reference Recordings. These are the recordings made by Ed Long and Ron Wickersham using their "Pressure Recording Process". The recordings are $12 each and can be obtained by writing Reference Records, P O Box 5046, Berkeley, CA 94705



RR-2 "Piano . . ."

We are pleased and very proud to present the first solo recording by the brilliant young Los Angeles pianist Steven Gordon. Mr. Gordon and his wife Nadya are a popular two-piano team who have made a number of recordings for the Klavier label.

Mr. Gordon studied for many years with the famous piano pedagogue Sergei Tarnowsky, teacher of Vladimir Horowitz. The impact of Professor Tarnowsky's special authority is abundantly evident in Mr. Gordon's playing of Chopin, which has drawn unanimous praise. The elusive character of Chopin's best music, the smallest phrase of which must be made to sing, is brought to exciting musical fruition in these performances by Steven Gordon.

An extensive search for a suitable instrument for this recording led us to the Yamaha CF Grand. A warm middle register, a moderately bright, ringing treble and an astonishingly powerful bass proved to be just the sound we were looking for. This appears to be the Yamaha's debut in a commercial recording.

The sessions were held in a seldom-used auditorium in San Francisco, which was originally built as a Druid meeting place, with the instrument not on the stage but on the floor of the hall. A great deal of experimentation was required before an acceptable balance was found between direct piano sound and the reverberant field.

The end product, we feel, is a highly realistic presentation of a particular piano in a particular hall. More importantly, this recording introduces to the world the exceptional artistry of Steven Gordon.

RR-3 "Percussion and . . ."

Kōtekan (pronounced ko-TEK-ahn) is an adventurous musical ensemble of six members, each of whom is a distinguished soloist. They come together mainly for fun, pooling their individual areas of expertise to create a unique repertoire and instrumentation.

Renee Grant-Williams is soprano soloist; Jon Lancelle plays acoustic string bass; Janet See performs on a variety of flutes; the three percussionists are Tom Hemphill, Richard Kvistad, and Jack Van Geem.

A Kōtekan concert is invariably full of surprises. The ultra-latest avant-garde fits happily between turn-of-the-century rags, parlor numbers, and original arrangements of more traditional music. For their first album, we have chosen a representative sample from their current touring repertoire.

The two vocal numbers could hardly be more different: La flûte enchantée from the song cycle "Sheherazade" by Maurice Ravel shows Ms. Grant-Williams to have totally mastered the subtleties of the French art song. "Lo! Hear the Gentle Lark" by Sir Henry Bishop is a Victorian period piece dashed off with brilliant coloratura abandon. The percussionists shine in "Liftoff" by Russell Peck, scored for nine bass drums, and "The Perilous Night" by John Cage, arranged for cake pans, coil springs etc. by Mr. Kvistad from the original score for prepared piano. The xylophone virtuosity of Tom Hemphill brightens "Rainbow Ripples" by George Hamilton Green, who was the foremost ragtime artist of his day. A world premiere recording of a piece specially written for the group by Richard Kvistad, "Dreaming of Another," brings together a dazzling display of the range and musical variety one can expect from Kōtekan.

RR-4 "Viola and . . ."

James Carter won his viola chair in the San Francisco Symphony at the age of 19, and for five years was its youngest member. He has produced two seasons of chamber music concerts which have been notable for high musical standards and for unusually adventurous programming.

The music for this first recording by the James Carter Chamber Ensemble is drawn from two concerts of the '76 season. One of their greatest successes was the Trio in E-flat, K. 498 of Mozart, featuring Philip Fath, first clarinetist of the San Francisco Symphony, and Sharon Mann Polk, distinguished pianist and teacher. This piece, subtitled "The Bowling Alley Trio," was written for a group of Mozart's friends whose enthusiasms included both music and bowling. James Carter and friends take this tip to bring us a performance of engaging informality.

For another concert, Mr. Carter brought in his sister and brother-in-law from Philadelphia. Deborah Carter is one of the most sought-after flute soloists on the east coast; her husband, William Smith, has for many years been keyboard player, assistant conductor and radio commentator for the Philadelphia Orchestra. From their concert together we chose the Duo Concertante in F for flute and viola by Hoffmeister (whom Mr. Carter calls "an obviously sadistic lunatic"), and the Trio Sonata in E minor by Loeillet in which the Carters are joined by Mr. Smith on harpsichord.

The tone of this recording is one of relaxed congeniality: an evening with a group of accomplished musicians who love to make music together.

WOMEN IN AUDIO

BODEN SANDSTROM

Casse Culver
Boden Sandstrom
(202) 332-4220

WOMAN SOUND

WOMAN SOUND
"Sound reinforcement & recording"
1735 New Hampshire Ave., N.W.
#104
Washington, D.C. 20009

WOMAN SOUND is a P.A. and recording company committed to quality sound reproduction. We offer:

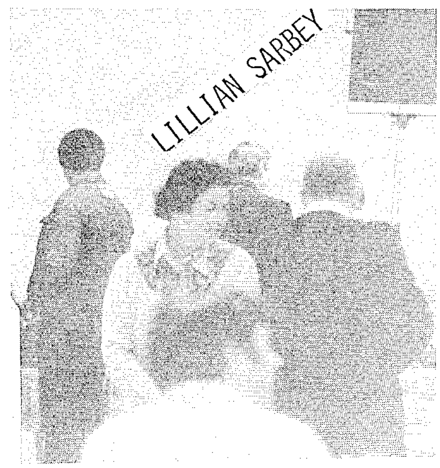
***SOUND REINFORCEMENT & ON-LOCATION RECORDING
We provide professional equipment and experienced engineers for any size concerts, meetings, demonstrations or clubs.

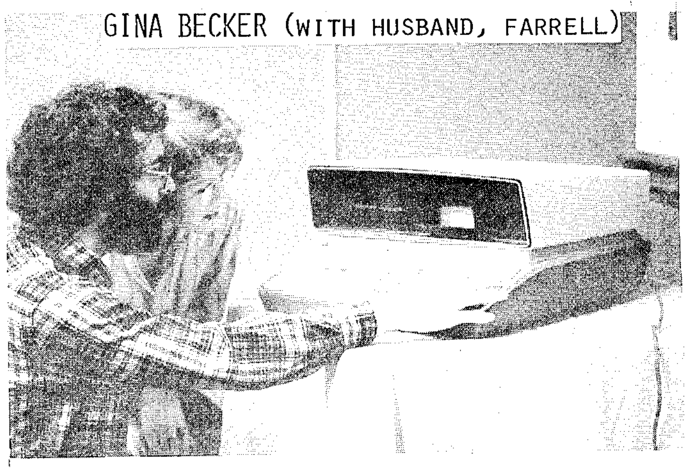
***STUDIO RECORDING
We record, mix and master up to 4 track demo tapes.

***RELATED SERVICES
We make, dub and edit tapes for cassette reproduction, radio broadcasts, dance concerts, and dances.

WOMAN SOUND has engineered concerts in such spaces as the Marvin Center Theater at George Washington University and other area colleges and clubs for performers such as Bev Grant, Holly Near and Cris Williamson. We have traveled with our skill and equipment to Baltimore, Philadelphia, New York, Atlantic City and Richmond. We have recorded solo artists, duos, and groups such as Casse Culver; Jeanne Mackey & Mary Trevor; LUCHA and THE BELLE STARR BAND.

WOMAN SOUND is owned and run by women and is dedicated to building opportunities for women in this highly technical and creative field. All our services are available to any individual or group. We believe you have the right to be heard clearly for a reasonable rate; therefore our low fees include the expertise of qualified engineers as well as the use of our quality equipment.





DISCO TURNTABLE ISOLATION

In recent classes the E.A.R. Corporation has asked us to pass out free "throw away" hearing protectors in exchange for a survey to find out who makes a good isolation material for turntables.

JERRY LAISERIN of E^{X}ponential Systems in Cranbury, N.J. sent us several companies making isolation material and added an interesting comment about isolating turntables. We have heard similar comments in class. Jerry writes:

Regarding the devices for Disco turntable anti-feedback, the latest address for Netronics R&D is 333 Litchfield Rd (Route 202), New Milford, CT 06776. Grado is still at the same old address. Another device I forgot to mention is a marble base from Discwasher (American Audioport, 1407 N. Providence Rd., Columbia, MO 65201). All of these have typical 4 Hz resonances and claim 20-40 dB isolation above that frequency. However, people-induced structural floor vibrations have a typical peak at 2 Hz. Essentially what is needed is a simple mechanical hi-pass filter with a sharp cutoff above 1 Hz. I'm reviewing some of the vibration work I did when installing electron microscopes in labs I designed for the Princeton U. Biology and Biochem Depts. If I come up with anything useful, I'll write it up for the Newsletter.

CALCULATING THE VELOCITY OF SOUND IN AIR

A knowledge of the exact velocity of sound when using Time Delay Spectrometry allows very precise *distance* measurements to be made by observing the frequency offset required to obtain the spectrum of a given reflection and then converting from frequency to time to distance.

The velocity of sound under conditions likely to be encountered in connection with architectural acoustic considerations is dependent upon three fundamental factors. These are:

1. λ is the ratio of specific heats and is 1.402 for diatomic molecules (air molecules)
2. P_S is the equalibrium gas pressure in N/M^2 (1.013×10^5 N/M^2)
3. p is the density of air in Kg/m^3

$$c = \sqrt{\frac{\lambda P_S}{P}}$$

where c is the velocity of sound in meters per sec.

The density of air varies with temperature and an examination of the basic equations reveals that, indeed, temperature variations are the predominant influence on the velocity of sound in air.

The equation for calculating the density of air is

$$\text{Density of air in Kg/m}^3 = \left(\frac{0.001293H}{(1 + [0.00367(°c)])76}\right)10^3$$

Where H is the barometric pressure in cm of Hg

°c is the temperature in degrees centigrade (Celsius)

$9/5(°c) + 32 = °F$

$5/9(oF) - 32 = °c$

Hg in inches times 2.54 equals Hg in centimeters.

Example

If we were to measure a temperature of 72°F and a barometric pressure of 29.92 in., we could proceed as follows: First we would calculate the density of the air according to the data gathered:

$5/9\ [(72) - 32] = 22.22°c$

29.92 in of Hg times 2.54 = 76 cm of Hg

$$\text{Density} = \left(\frac{0.001293(76)}{1 + [.00367(22.22)]76}\right)10^3 = 1.1955 \text{ Kg/m}^3$$

and, having made the metric conversions and obtained the density figure, we can then use the basic equation for velocity

$$c = \sqrt{\frac{1.402\ (1.0\ 3 \times 10^5}{1.1955}} = 344.67 \text{ m/sec}$$

Since we started with the dimensions commonly used here in the United States, we then convert back to them by

$$\frac{344.67\cancel{m}}{1 \text{ sec}} \times \frac{100\ \cancel{cm}}{1\ \cancel{m}} \times \frac{1\ \cancel{in}}{2.54\ \cancel{cm}} \times \frac{1 \text{ ft}}{12\ \cancel{in}} = \frac{1130.81 \text{ ft.}}{\text{sec}}$$

ENOUGH TO MAKE A STRONG MAN WEEP

Shattering Disclosures

In a recent TV commercial, singer Ella Fitzgerald is shown tape-recording a high note that shatters a wine glass when the tape is later played back. The commercial's highly effective message is that this particular brand of tape reproduces voices with unparalleled fidelity.

But now comes a spoilsport, Adrian Hope, of the British magazine *New Scientist*, who is trying to knock down the whole notion that Ms. Fitzgerald or any other singer can shatter a glass with sheer lung power. "I have devoted days to searching the literature," Hope says, "and I could never find any reference to this."

In the end, the secretary of Britain's Advertising Standards Authority has backed away from his earlier support for the commercial and has admitted that despite the many stories about Caruso and others practically vaporizing champagne glasses with their voice boxes, "The evidence [for such a feat] is purely hearsay."

Does this really mean that sound can't splinter a wine glass? Not at all, says Hope. You can splinter all the glasses you want to, so long as you have a loudspeaker system that can produce sound levels 10 decibels greater than those the human ear can endure. But when that level is reached, Hope explains, "You're in the area of mixing cement [by sending sound waves through it]—something which is actually done in the U.S. This is hardly the domestic setup shown in the ad."

SR has a section of the magazine in which they reproduce news items from other publications.

"Shattering Disclosures" appeared in a recent issue. Pete Tappan of the consulting firm, Kirkegaard & Assoc., and grad of our Chicago 1976 class, is responsible for the famous Memorex "shattering glass" ads on TV.

Several years ago Pete gave a beautiful paper at the Los Angeles AES in which he demonstrated with recordings that the human voice could easily break a glass at a very moderate sound pressure level.

It isn't at all difficult to determine the resonant frequency of a particular glass that will cause it to shatter.

GROUND LOOP IMPEDANCE TESTER

Having now used the Woodhead GLIT for almost six months, we are pleased to report that it is even more valuable than we first thought it would be.

When shown to many engineers, especially broadcast engineers, their first question is "How much over $1,000 is it?" The less than $200 price seems hard to believe.

The GLIT has proven capable of revealing when only a common is available rather than a true ground.

It is ideally suited to test between chassis and ground in musician's systems.

The accompanying instruction book contains much useful information and fully describes the standard applications of this device.

Two useful equations to keep in mind while using the GLIT are

$$\Omega_{max} = \frac{120V}{5 \times I \text{ rating}}$$

and

$$I_{max} = \frac{120V}{5 \times \Omega}$$

For example, if you were on a 20 amp, 120V circuit, then the *maximum* impedance you would want to see would be

$$\Omega_{max} = \frac{120V}{5 \times 20} = 1.2\Omega$$

Or, if you measured an impedance of 2.0Ω, then the maximum current should not exceed

$$I_{max} \frac{120V}{5 \times 2.0} = 12 \text{ amps}$$

Daniel Woodhead Company, 3411 Woodhead Drive, Northbrook, Ill 60062

Safety Yellow® G·L·I·T

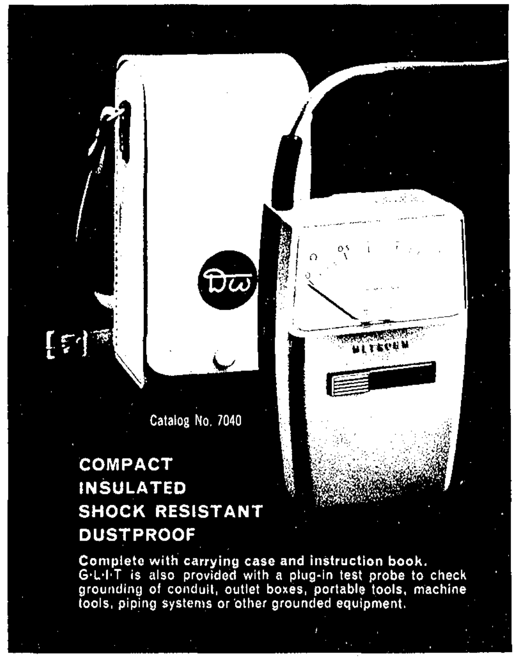
GROUND LOOP IMPEDANCE TESTER

BE SURE OF YOUR GROUNDS . . . EVERY TIME . . . EVERYWHERE . . . WITH THIS COMPACT METER!

Now, for the first time, you can conveniently check your circuits to make certain low impedance exists. The Woodhead SAFETY YELLOW Ground Loop Impedance Tester measures the quality of the ground and clearly registers the result in ohms on a direct reading scale.

Unlike all other testing devices, the G·L·I·T tests a live circuit when it is hot. It actually sends through the ground, from the power source itself, a momentary surge of about 20 amperes (too quickly to actuate the overcurrent protection!) The resulting impedance measured by the tester is then shown on a scale in ohms.

You no longer need only *hope* that the ground is good enough to protect equipment and personnel. Now you can actually measure the quality of your ground circuit and periodically recheck the condition of your grounds as a part of your continuing program of preventive maintenance.



What Is A Good Ground?

To get technical for a moment, the ideal ground on a 20 amp. circuit should have an impedance (AC resistance) of no more than 1.2 ohms. Why? The ground path should permit five times the rated current to flow. For example, 120 volts ÷ five times the rated current (in this instance, 5 x 20 amp., or 100 amps) results in 1.2 ohms as the permissible impedance. Your Ground Loop Impedance Tester, used in conjunction with this simple formula, tells you immediately if you have a low impedance ground.

How Does G·L·I·T Work?

The Woodhead GROUND LOOP IMPEDANCE TESTER draws its power from the actual circuit under test. G·L·I·T electronically simulates a fault condition in the line-ground circuit, causing a fault current (short) to flow through the circuit at about 20 amperes for 1/25 of a second. This fault current is of such short duration that it will not actuate the overcurrent protection. The current flow is then measured electronically, and the results indicated in ohms on a direct reading scale.

How G·L·I·T Is Used

It couldn't be easier! Just . . .

(1) Insert plug into standard grounding type receptacle.

(2) Press operating button and read impedance from scale. Meter maintains reading as long as the button is depressed.

SPECIFICATIONS

Voltage Range: 90 to 130 V. AC, 50/60 cycles H_z.

Test Current: Approx. 21 amps with a loop impedance of 1 ohm at 120 V.

Size: 5″ x 3¼″ x 2″.

Weight: 2 lbs.

Construction: Shock resistant taut band suspension meter in an ABS plastic case, specially designed for portable use.

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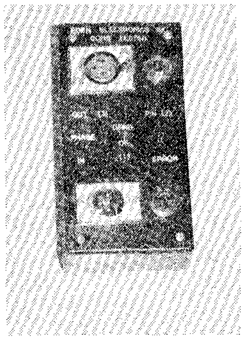
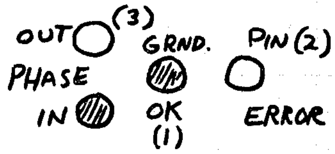
CABLE TESTER FROM EDEN ELECTRONICS

DAVID NORDSCHOW, president of Eden Electronics and a graduate of our 1977 Minneapolis class, recently sent us a true "black box" cable tester. Dave showed us a prototype during the Minneapolis class and said that he hoped to be able to produce the unit for Syn-Aud-Con graduates for $19.95. We told him that if he could get it into production for that price, he had a market.

To quote Dave's recent letter, "They are in stock and ready for delivery. The list price is $29.95. They are available to Syn-Aud-Con graduates for $19.95 plus $1.00 handling and postage, in lots of four or more they are $15.00 plus postage.



The tester has XLR type connectors and phone plug connectors. Four LEDS automatically light to describe the status of the conductors of the cable.



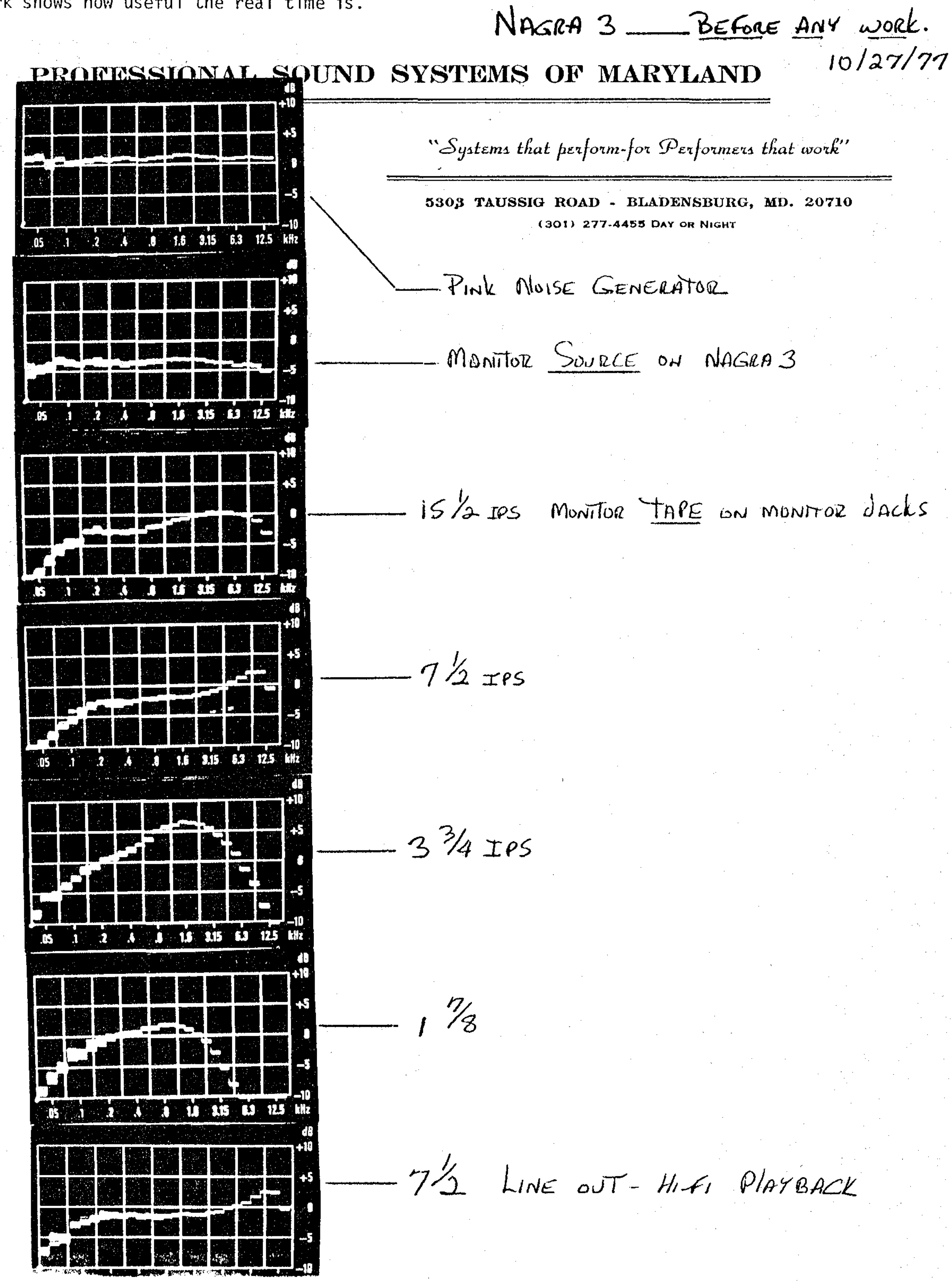
When "in" and "ok" light, all is well. The others are self-explanatory.

We have found the unit rapid, accurate and very handy to carry and use.

Write David Nordschow, Eden Electronics, 2839 Johnson St., N.E., Minneapolis, Minneapolis 55418.

SERVICING TAPE RECORDERS WITH A REAL TIME

AL LAKOMYJ of Professional Sound Systems of Maryland and a graduate of all of our 5 DC classes used his Altec 8050A and the Polaroid CR-9 Oscilloscope Camera to do some excellent service work on a tape recorder. A glance at his work shows how useful the real time is.



GOING PRICE ON THE HP 97

ALAN GERMAN, Sales Manager at Superior Sound in Wichita and a grad of our 1977 Kansas City class, purchased an HP 97 right after class. He wrote "Concerning the HP 97, Olympic Sales (in Los Angeles) was *very cooperative*. Ordered by phone on Monday, received the unit that Friday via U.P.S. They shipped C.O.D. and the cost of the unit was $600. Needless to say, I was quite pleased with the price."

1/3-OCTAVE SPACINGS AND THE dB

Interesting questions can lead to new insights into old answers, especially if each problem is approached for its most general case solution. Just that occurred in answering a question TIM GUHL of the Syracuse Civic Center and two-time graduate, asked during the 1977 Boston class. Tim asked why 1/3-octave spacings followed one dB steps.

Whenever the metric community is faced with an idea conceived in the English system which they would like to adopt they adapt it to the nearest decimal form. It took some metricationist the merest flash of time to observe that a true 1/3-octave series based on

$$2^{1/3} = 1.259921050$$

could be as easily expressed by a decimal exponent of the base 10

$$10^{.1} = 1.258925412$$

and from this observation, a series of preferred numbers called Renard numbers and preferred labels were constructed by one of the international committee.

It was merely coincidental that 1 dB happened to be a power ratio of

$$10^{1/10} = 1.258925412 \text{ also}$$

The 20 series Renard numbers for 1/6-octave filter spacing coincide with voltage ratio steps since

$$10^{1/20} = 1.122018454$$

is 1 dB expressed as a voltage ratio.

The preferred labels are simply convenient rounding off of the exact preferred numbers. The exact preferred numbers provide even logarithmic spacing per decade for each filter.

A further expansion of this idea is developed as follows: Suppose you wished to plot a response curve from 10 discrete measurements between 50 Hz and 10,000 Hz and wished each point on the chart to be equally spaced (since frequency response charts are normally logarithmic, we need even logarithmic spacing.)

$$LFX \times \left(\frac{UFL}{LFL}\right)^{\left(\frac{1}{N-1}\right)}$$

is the general equation

Where LFL is the lower frequency limit in Hz
UFL is the upper frequency limit in Hz
N is the number of plots desired

Therefore,

$$\left(\frac{10{,}000}{50}\right)^{\left(\frac{1}{10-1}\right)} = 1.801648$$

and

$$50 \times 1.801648 = 90.08 \times 1.801648 = 162.30 \text{ etc.}$$

So that ten equally spaced plots on a logarithmic frequency scale would be at

Plot #	Frequency	Plot #	Frequency
1	50 Hz	6	949.12 Hz
2	90.08 Hz	7	1,709.98 Hz
3	162.30 Hz	8	3,080.78 Hz
4	292.40 Hz	9	5,550.47 Hz
5	526.81 Hz	10	10,000.00 Hz

ANTIQUE ELECTRONIC CALCULATOR

FARRELL BECKER, Head Soundman at Wolf Trap and 3-time graduate, brought in a $5 electronic calculator to the 1977 DC class. It turned out to be an early Frieden unit that originally sold for well over $1,000. Farrell picked it up at a garage sale. It adds, subtracts, multiplies, divides, and I believe it does square roots also. A truly valuable historical collector's item. (See Newsletter Vol. 3, No. 2 for a write-up of Farrell's garage sale purchase - Magnavox horn for $25, an antique prize.

THE D_C MODIFIERS - M_A AND M_E

The architectural modifier of D_c, hereafter referred to as M_a, occurs when almost all of the sound energy emitted by the loudspeaker encounters a highly absorptive area large enough to materially reduce its power before the energy has had any opportunity to be reflected by any other surface. Figure 4-20 in *SOUND SYSTEM ENGINEERING* illustrates a limiting case example in (A) on page 69. Since there are no perfect loudspeakers and no perfect absorbers, the M_a value is usually a very low multiplier.

Theory of M_a

This D_c modifier M_a results from the removal of *additional* energy from signal emitted upon its first encounter with a selected absorbent boundary surface than would have been expected had the same energy first encountered a surface possessing the average absorption coefficient of the space as a whole.

In the limiting case, if the area the sound energy first encountered were 100% absorptive and if none of the energy encountered any other surface, $Q \equiv C_\angle \equiv$ area of absorption, then

$$M_a = \frac{1-\bar{a}}{1-1} = \infty$$

Which can intuitively be acknowledged in this case as well.

If, instead, the first surface area encountered had an absorption coefficient the same as the average absorption coefficient ($\bar{a}$) for the space, then $(1-\bar{a}) \equiv (1-a_c)$

$$M_a = \frac{1-\bar{a}}{1-a_c} = 1.0$$

Thus, the effect disappears.

It would appear that M_a is only of interest when

1. $Q \equiv C_\angle$ or nearly so
2. $a_c >> \bar{a}$

Whenever $Q \not\equiv C_\angle$ then the ratio of the perfect Q for a given *point* ($Q_{theor.}$) receiving the same power as all other points covered to the Q actually used $Q_{act.}$ is accounted for in the equation

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

In real cases $Q_{act} << Q_{theor}$ leading to M_a remaining a relatively small multiplier.

It is worthwhile at this point to consider what is actually happening at a listener's ears if he is situated on the absorbing surface that the energy first encounters. The higher the M_a the higher the ratio of direct-to-reverberant sound he hears. The main purpose of increasing Q or M_a is to increase this ratio at the listener's ears. Another parameter available to us that allows us to accomplish the same results is to *move the loudspeaker closer to the listener*. If I move a loudspeaker of any given Q in a room of any given M_a to one-half its former distance from the listener, I raise the direct-to-reverberant ratio at his ears by plus 6 dB. This could also be accomplished by leaving the loudspeaker at its original position and raising its Q by a factor of four times.

This parameter, the distance from the loudspeaker to the listener, D_2, can be handled in the following manner.

1. Determine the Q_{AL} value necessary for the D_2 at hand according to the $\%AL_{cons}$ equation. Call this D_2 the D_{2SS}, meaning the distance from the loudspeaker to the listener if the array were a single speaker (SS)
2. Then check to see if $Q_{avail} = Q_{AL}$ and if that is possible, see if the Q_{avail} matches the required $C_\angle$. If it does not, then double the Q of the first speaker and add a second speaker with a Q = to ¼ of that now assigned to the first speaker.

   If coverage is still not attainable, then triple the Q of the first speaker, make the second ¼ the Q of the first, and the third speaker ¼ the Q of the second speaker. Again check coverage required vs $C_\angle$ available.

Suppose at this point that the first speaker's Q requirement is now higher than any Q_{avail} because of the N factor that has developed by adding other loudspeakers to the array. Our recourse is to find an electroacoustic D_c modifier, M_E that will exactly cancel the detrimental influence of the N factor. M_E can be found by

$$M_E = \frac{NQ_{AL}}{Q_{avail}}$$

Where N is found by the method described in Tech Topic, Vol 5, No. 5 by Ed Lethert

Q_{AL} is the Q required by a single speaker system to reach D_{2SS}

Q_{act} is the Q of the device actually available to you

The new distance, D_x, that you will place this loudspeaker from the listener is found by

$$D_x = \frac{D_{2SS}}{10^{\left(\frac{M_E}{20}\right)}}$$

DC MODIFIERS, CONT.

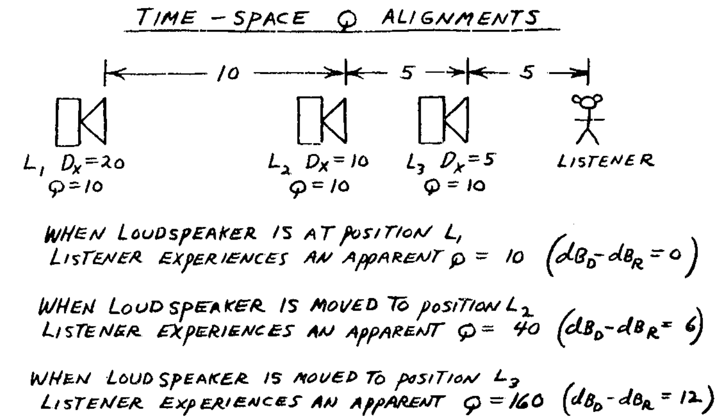
Digital time delay now adjusts this loudspeaker so that its signal, so far as the listener is concerned, is acoustically part of the original array.

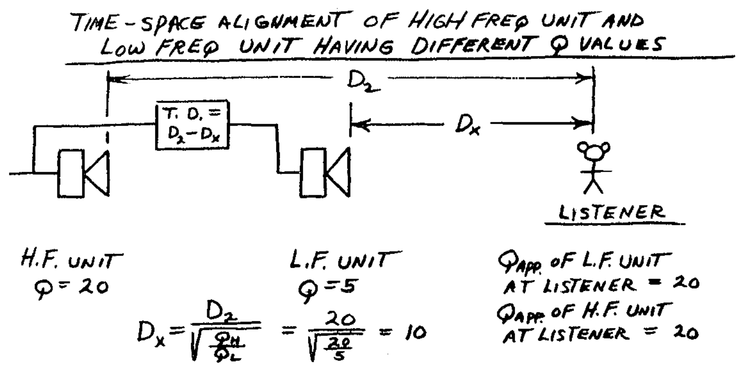
The Design Procedure Summarized

1. Find Q_{AL} from Peutz' $\%AL_{cons}$ equation
2. Examine coverage and increase units as needed and assign Q_{avail}
3. Calculate N for the array that provides coverage
4. Adjust D_x to cancel N with M_E for those loudspeakers which cannot meet the required Q necessary to operate over the distance D_{2SS}
5. Adjust time delay to the equivalent of $D_{2SS} - D_x$

Whenever possible we try to follow the philosophy that

1. One loudspeaker is best if it can both meet the Q requirements and the coverage needs
2. The first compromise is more than one loudspeaker in the same location but aimed at differing coverage areas
3. The next compromise is loudspeakers operating at different physical locations but acoustically from the same point.
4. Maximum compromise is high density overhead or down one wall distribution.





By working through the steps described above you arrive at each compromise with the assurance that it is not possible to meet the Q and coverage requirements by any lesser compromise.

Summary

Array design undertaken with these tools leads to highly intelligible and intelligent arrays. Only acoustic gain considerations need then be checked before bringing the array design to a successful conclusion.

Note that digital time delay as used here should be exact time alignment. The so called "Haas Effect" is then applied to the *whole* array not to the parts of the array.

ARTICLES OF INTEREST

The IEEE SPECTRUM for December 1977 contains an article by Harold S. Black entitled, *Inventing the Negative Feedback Amplifier*. Mr. Black, 79 years young, has written in some detail how "six years of persistent search helped the author conceive the idea 'in a flash' aboard the old Lackawanna Ferry."

It was on August 2, 1927 that Black conceived the original idea for a negative feedback amplifier. "Although the invention had been submitted to the U.S.Patent Office on August 8, 1928, more than nine years would elapse before the patent was issued on December 21, 1937 (patent No. 2101671)."

One reason for the delay was that the concept was so contrary to established beliefs that the Patent Office initially did not believe it would work. That it did indeed work was due to the remarkable "flash insight" that an amplifier's output fed back to its input out of phase would not oscillate if at any frequency from zero to infinity the loop transfer factor were not allowed to become real, positive, or greater than unity.

Just as Harold S. Black's original papers are collectors items, so this excellent, informative reminiscence can be beneficial reading.

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ALEX ROSNER, president of the well-known Rosner Custom Sound and 4-time graduate, and Larry S. King of KMK Associates, acoustical consultants, have written an excellent, detailed, and instructive paper on the *New Mobile Sound Reinforcement System for the Metropolitan Opera/New York Philharmonic Orchestra Park Concerts* in the September 1977 issue of the J-AES, Vol 25, no. 9, pages 566-571.

This is one of those rare systems-description articles where both authors are well qualified, willing to share their knowledge and orderly in doing so. The loudspeaker cluster design is shown in detail and reflects the usual high standards of KMK Associates.

BOOKS OF INTEREST

On rare occasion I find a highly desired edition of some long sought book staring me in the face from a second-hand bookdealer's dusty shelf.

During our annual stopover in Gettysburg, PA, I found myself eyeball-to-eyeball with the *MATHEMATICAL HANDBOOK FOR SCIENTISTS AND ENGINEERS* by Korn and Korn, published by McGraw-Hill. This 943-page handbook is not for everyone. It is an invaluable reference, however, for those of you actively studying mathematics via computer programming, etc.

Among this volume's virtues are excellent chapters on "Curvilinear Coordinate Systems - Chapter 6; "Real and Complex Numbers - Elementary Algebra - Chapter 1 which contains the following definition:

$$C^x = a \quad \text{or} \quad c^{\text{Log}_e a} = a$$

and a further definition that

$$\text{Log}_c c = 1$$

thus we can write our basic definition of an exponential expression in logarithmic form as

$$\frac{a}{c} = b^n$$

or $$\text{Log}_b \frac{a}{c} = \log_b b \times N$$

and since $\text{Log}_b b = 1$

we then have $\text{Log}_b \frac{a}{c} = N$

It is this kind of mathematical shorthand that crams the entire book full of an unbelievable quantity and quality of information. Out of the 62 math books currently on my bookshelves, including the giant *HANDBOOK OF MATHEMATICAL FUNCTIONS* by Abromowitz and Stegun, published by The National Bureau of Standards, I am unable to find as useful and insightful definition.

There is a really thorough chapter on "Probability Theory and Random Processes, Chapter 18, which includes mathematics of cross correlation, spectral densities and ergodic random processes. This book left me in a humbler mood but infinitely more curious to learn more about the vast storehouse of mathematics.

I paid $5 for my used copy of *MATHEMATICAL HANDBOOK FOR SCIENTISTS AND ENGINEERS*. Published by McGraw Hill, 1961.

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Edwin Howard Armstrong invented the regenerative, super regenerative, and superheterodyne AM radio receivers. Then as a crowning accomplishment after having made AM broadcast radio possible in the first place, he invented a completely new system of broadcasting and presented it "full blown" to the world - FM radio.

There literally would have been no radio or television broadcasting without these basic discoveries. Major Armstrong became one of the wealthiest men in the United States as a result of his AM inventions. When the economic-political strength of the large corporations (who had used his earlier inventions to gather much of their power and had been forced to pay him for them) were faced with Armstrong's latest invention, FM radio, they turned in combination on him in a struggle to the death.

A MAN OF HIGH FIDELITY - EDWIN HOWARD ARMSTRONG by Lawrence Lessing originally published by J.B. Lippincott Co (also available in paper back version), describes and documents the tragedy that stalks the individual inventor when he seeks justice in the courts in his battle with giant corporations.

The actions of Brigadier General David Sarnoff of RCA, as described by Lessing, are corrupt and manipulative; Lee DeForest as crassly ignorant and stupid; and the deadly skill of the corporate teams of lawyers is chillingly portrayed as they use the senility of the politically oriented Supreme Court Justices to pervert completely the course of justice. It's a lesson in how effective public relations personnel are in sugar coating pirates and barbarians in modern life.

Armstrong stood as a lone knight in a rare order of nobility against the need of academia and industry to crush the very integrity of a great mind. They sought to rob him of recognition or even existence of his work. This well-documented, accurate, and important book should be read by anyone concerned about the so-called "quality of life" in the free world. The kind of "professional management" that fought Armstrong and eventually drove him to suicide would do the same to any individual they detect as an obstacle in their path.

Large enterprises are not inherently evil nor are all individuals virtuous. In our increasingly technological society the questions posed by this book and its minute description of the tragedy of one genius cry out for better answers than now available. Answers that allow order, organization, and advancement for our material society while protecting, honoring, and rewarding the creative thinkers capable of triggering such advances.

(continued next page)

BOOKS OF INTEREST

CONTINUED

BOBBY GOODMAN, president of Associated Sound Inc in Annandale, VA and two-time DC graduate, recommended that we obtain *ELECTRICAL CODE DIAGRAMS* by B.Z. Segall.

To anyone asked to deal with standard electrical codes, this complete reference work is extremely useful. Every conceivable interconnection of any imaginable electrical device is touched upon in this two volume (ring binder) voluminous reference. Invaluable for use in illustrating what the electrician should do and in defining terms so the electrician and the audio engineer use the same vocabulary with regard to the electrical power system.

These volumes are expensive ($45), well prepared and printed in a serviceable size and clarity. Can be ordered from Peerless Publishing Co., P. O Box 30187, New Orleans, LA 70190.

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BEGINNERS GUIDE TO MICROPROCESSORS, Tab Books, $5.95 paper back ISBN 0-8306-6995-7 by Charles M. Gilmore

This appears to be a good *basic* book to let a newcomer learn about the field and get started in a modest way. Well illustrated and fundamental, it is not difficult reading and it leads in a practical way to considering using your own unit.

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THE ILLUSTRATED DICTIONARY OF BROADCAST - CATV - TELECOMMUNICATIONS by R. Terry Ellmore, Tab Books #950, paperback, ISBN 0-8306-6950-7. $8.95.

You'll find definitions here that are hard to find elsewhere:

ANTARA: Jajasan Kantorberita Nasional Antara. An Indonesian news agency.

AO: Audio operator. The individual who operates an audio control console

NAGRA: The Polish word for tape recorder. A high quality tape recorder designed by Stefam Kudelski and manufacturered in Switzerland.

Depth: One of three dimensions. The others are height and width.

Womp: A momentary surge of power in a television receiver that results in a corresponding increase in picture brightness.

Guonking: Noise or movement that is not connected with a program, but is picked up by a microphone or camera.

While not the most avant-garde technical information, the vocabulary of the news business, advertising agencies, announcers, etc. all are included. Subjects covered range from acting, film lighting, history, and national and international organizations that are relevant to the broadcasting industry.

CLASSIFIED

FOR SALE:

Hewlett Packard HO 1-8056A real time 1/3-octave audio frequency analyzer. This model requires connection to your oscilloscope. Generates a real time display on your oscilloscope from 63 to 12,500 Hz with a 20 dB vertical scale. The HO 1-8056A originally sold for $3610. and represents an excellent buy at $2500. We have seen this unit and it is in excellent condition.

Contact: Steve Simpson, Jr., Southwest Sound, 2323 Loop 410 N.W., San Antonio, TX 78230. Ph 512/541-4411

FOR SALE:

GR 1523 Graphic Level Recorder with a 1523-P4 Wave Analyzer plug-in unit. This combination allows 10 Hz bandwidth, 80 dB dynamic range, linear or log display, 10 Hz to 80 KHz frequency range analysis with a tracking output. This is a deluxe way to measure and record detailed distortion spectrums, filter responses. The equipment is in mint condition.

Contact: John Odum, The Music Mart, Pennyrile Mall, Hopkinsville, KY 42240. Ph 502/885-5386

WANTED:

Altec passive filters 9013-16 (2000 Hz) and 9013-17 (2500 Hz). Needed to complete a set of 9014.

SELL OR SWAP:

Altec passive filters 9013-10, -13, -14 (500,1000, & 1250 Hz).

Contact: B. Martin, Central Jersey Sound Center, P O Box 332, Oakhurst, N.J. 08855. Ph. 201/542-4100

FOR SALE:

U.R.E.I. 100A Sonipulse without mic	$650.00	(Used test equipment in excellent condition)
Communications Co. RT-60 Reverb Meter	300.00	
H.H. Scott type 450 Sound Level Meter Just factory reconditioned	150.00	

Contact: Jerry Marshall, Howell Electronics, 2873 Pershing Dr., El Paso, TX 79903. 915/566-3968

FOR SALE:

Altec-HP 8050-A real time analyzer (the 8050A's are a rare item on the used market)

Contact: Rick Garner, Audio Services, 620-15th Street, Moline, IL 61265. Ph 309/797-9891

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