SYNERGETIC
SYN AUD CON
AUDIO CONCEPTS

newsletter

VOLUME 8, NUMBER 3
SPRING, 1981

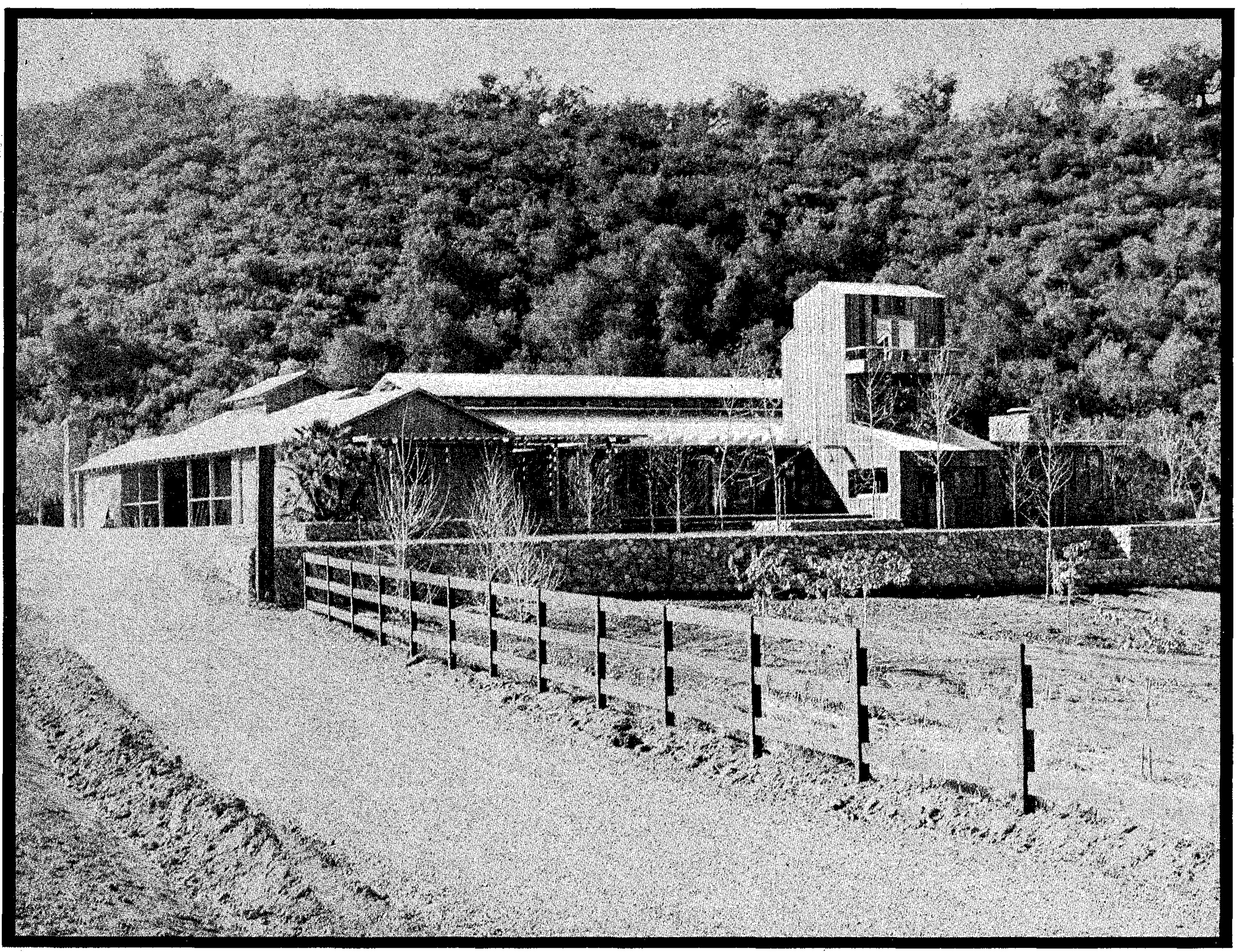
P.O. Box 1115, San Juan Capistrano, CA 92693
Ph: 714-496-9599

SYNERGETIC
Working together; co-operating, co-operative

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

EXCHANGE OF IDEAS

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



SYN-AUD-CON AUDIO INDUSTRY SEMINAR CENTER

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SYN-AUD-CON EDITORIAL

Syn-Aud-Con classes have traditionally provided exceptional companionship among the participants--so much so that Syn-Aud-Con is more of a society or fraternity than it is an educational institute.

We have always been eager to share our love of audio and our excitement in its future.

Graduates who have attended one of our new classes at our Seminar Center tell us that we now have the clearest, most concise presentation of audio fundamentals available anywhere.

We believe them because we have spent the past year constructing new, powerfully presented, totally accurate demonstrations of the audio fundamentals. The same fundamentals that we are seeing "professionals" in our business stumble over again and again. Current popular magazines on audio have become so in error as to constitute a sub-society of out-of-phase thinkers who feed each other their false concepts.

We now have a "hands-on" demonstration of "gain"; an exciting demonstration of UREI's new amplifier that corrects for error at the loudspeaker; new insights into log and linear scaling, decades, octaves; and rapid new ways to handle exponential notation in both phase and impedance measurements.

Participants in the first five classes to be held at the new West Coast Seminar Center make a common observation. "No picture can begin to tell anyone what you have here," and, "I never expected anything of this magnitude from the pictures," are frequent remarks. It sounds self-serving to say that this is a magnificent facility but that's what we are being told by those who have seen it in person.

We are eager to share what we truly feel is an environment that reflects what life is really all about. One needs to escape the tawdry mesmeric world of city and urban pressures and influences and have a chance to *feel* what our forefathers' earth was like. We all know that "left" brain learning and discipline can prosper when the "right" brain "gut feeling" is freed from the physical and mental pollution of this age. The Seminar Center is an educational insight into a great deal more than audio.

The Newsletter is now being offered for the first time to non-Syn-Aud-Con graduates (at a higher price than graduates pay), and we are receiving substantial numbers of new subscribers. For them, we will be running in subsequent issues at least one tutorial article per issue.

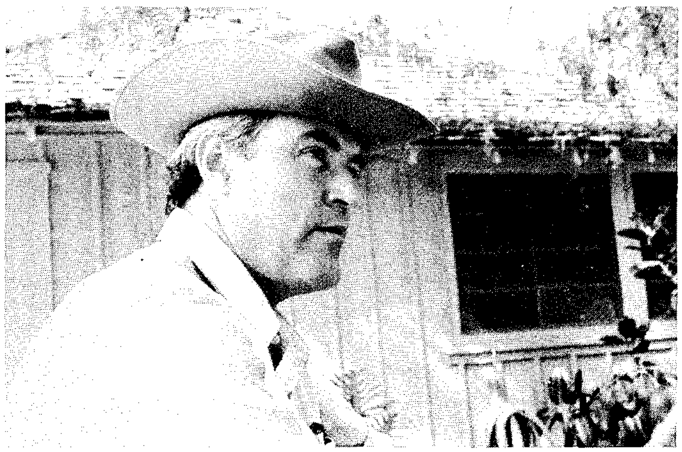
We believe that a year's exposure to our Newsletters and Tech Topics is an excellent preparatory step towards receiving full benefit from attending a Syn-Aud-Con class. Our philosophy is harder to reduce to tutorial articles but we hope our editorial content hints at it just a little.

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

SYN-AUD-CON STAFF

Jeanie Garcia

Jan Kreitz



Don Davis

Carolyn Davis

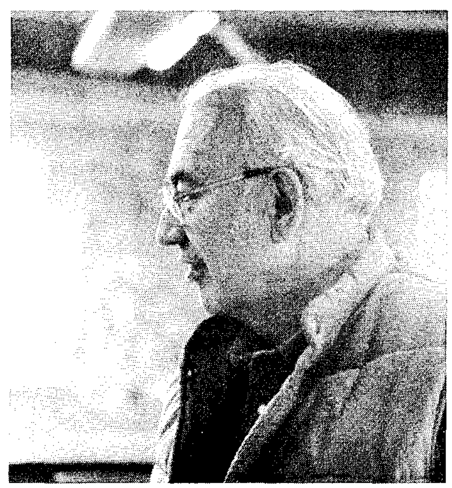
When you call Syn-Aud-Con on a day-to-day basis, one of four people will answer the telephone:

Jan Kreitz	Don Davis
Jeanie Garcie	Carolyn Davis

For those new subscribers to the Newsletter, here are pictures of all four taken during a recent Syn-Aud-Con seminar.

We all enjoy seminar time and the chance to get to know those who attend. We find that they, too, are eager to get to know the people they have talked to on the telephone.

Just prior to, during, and directly after seminars, Ken Wahrenbrock provides extremely valuable assistance to Syn-Aud-Con. Those wishing to reach Ken at such times can do so through Syn-Aud-Con.



Ken Wahrenbrock

WELCOME NEW NEWSLETTER SUBSCRIBERS

Last Fall at the AES Convention we announced that the Newsletter subscription was open to anyone who wished to subscribe at $50 per year for non-graduates.

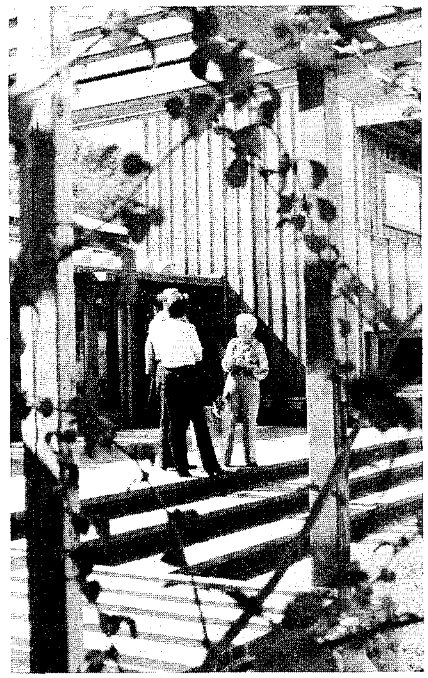
HARVEY EARP of J. W. Davis & Company reproduced the Newsletter offer in their "Jay's Jargon" publication recently.

We are very pleased to welcome many, many new subscribers as the result of their mailing. Much of the Newsletter is written with the assumption that the subscriber has attended a Syn-Aud-Con class and knows all the "alphabet soup," PZM, LEDE, TDS-ETC, TEF, ad infinitum. Therefore, a few back issues of the Newsletter are included with the first mailing to a new subscriber.

LEARNING IS FUN

Few realize it until they experience it, but Syn-Aud-Con coffee breaks, lunches and dinners are fantastic learning times.

Here you are gathered together with some of the most advanced professionals in audio, totally relaxed and discussing subjects of mutual interest, triggered by the often startling demonstrations you have just witnessed in class.

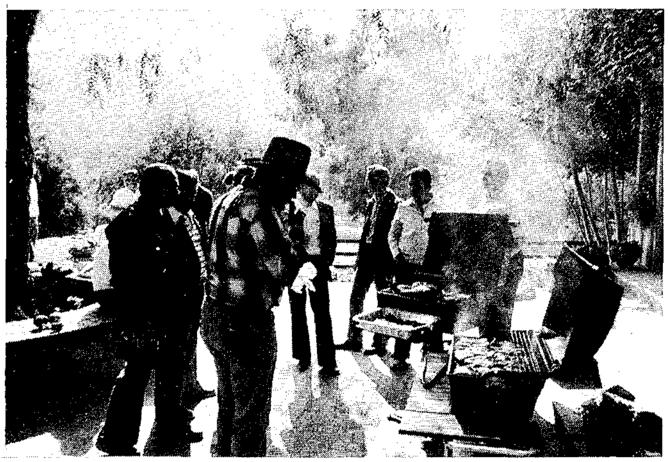


Coffee breaks (beverages, doughnuts, fruit, etc.) are held around the large stone fireplace or out on the front entrance patio or around one of the reflecting pools.

Lunches, on warm days, are held on the large patio at the old main ranch house and on cool days inside the ranch house or the Seminar Center itself.

Lunches are abundant buffets that encourage you to gain weight as well as knowledge.

Dinners are something special at the West Coast Seminar Center. The lights of the L.A. basin are visible 60 miles away on clear nights as we go to dinner. We have a truly sumptuous substantial meal prepared by a skilled catering service. We can assure you that one of the unexpected benefits of our west coast classes is the escape from "motel menus."

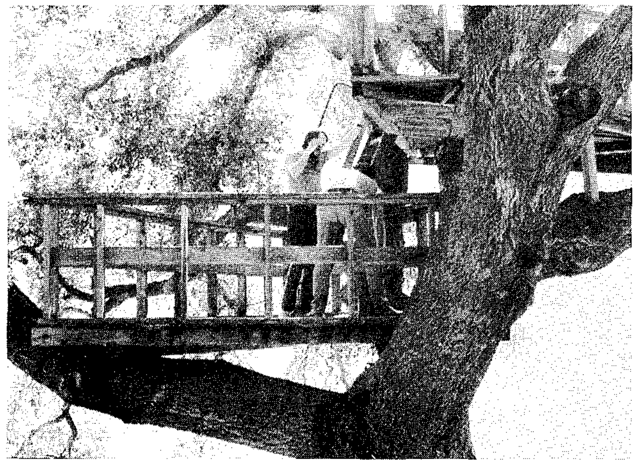


The last day of class the noon meal is a steak barbecue with Rancho Carrillo Beans to compliment the really man-sized steaks that Jerry Kreitz (Jan's husband) prepares just for you. Special salads, baked potatoes and abundant beverages complete the story.

We're proud that our demonstrations and "hands-on" excercises can and do keep you awake even though full and content.

4,000 graduates in the past eight years have repeatedly told us that Syn-Aud-Con is an audio experience, not just instruction. We see no reason not to make it a culinary one as well.

A quick swim, a game of tennis, exploring a tree house, or a snooze in a hammock between two trees and you're ready to handle whatever the rest of the day brings.



AVAILABILITY OF THE CROWN DEDICATED TEF™ INSTRUMENT

During the recent (February, 1981) TEF™ session with Dick Heyser, we heard from Crown personnel regarding the many exclusive features their instrument will possess relative to any existing equipment: time delays of 200 *secs* - Yes! *secs*; full phase measurements - both absolute and relative; polar or frequency plots. Crown's instrument utilizes several never-before-used techniques that will quickly convince any user of their superiority in this field.

Availability? Anyone's guess is as good as ours. Our best guess (emphasis on *guess*) is that end users will first see this remarkable instrument in early 1982. Heyser and Syn-Aud-Con will be very pleased if a prototype passes into our hands any time this summer.

We accepted the delay in Crown's completion of the prototype philosophically. We'll not be as philosophical if they overrun their goal of $8000 for this instrument. We suggest you let them hear from you regarding the importance of holding this price.

We will continue to keep you informed. Remarkable accomplishments in TEF™ measurements are occurring daily using available rental equipment and the Syn-Aud-Con/Heyser module. While we continue to suggest that current licensees not purchase existing equipment unless its amortization is a certainty within six months, we do encourage you to rent the equipment every time the opportunity to cover the cost of the rental arises.

WANTED HP 21

Syn-Aud-Con needs to add to our stock of HP21s. We will pay $25 for an HP 21 (in good condition), power supply and instruction manual. Call or write Syn-Aud-Con soon if you want to sell one.

SYN-AUD-CON 1981 SCHEDULE

JUNE 23-25	**SEPTEMBER 22-24**
JULY 21-23	**OCTOBER 20-22**
AUGUST 18-20	**NOVEMBER 17-19**

NEW PRICE SCHEDULE

The Syn-Aud-Con registration fee scheduled 2 years ago at $500 will increase to $600 U.S. funds starting July, 1981.

Registration fee for Syn-Aud-Con graduates will be $525.

Registrations for 1981 classes July through December paid in full prior to July 1, 1981, will be at the current fee of $500 ($450 for graduates).

The Newsletter subscription will be $30 in the United States and $35 U.S. funds outside the United States. (The $25 charge for the Newsletter was established in 1973.)

SPECIAL SYN-AUD-CON WORKSHOPS

One thing is definite: Syn-Aud-Con graduates want to attend special workshops. The response to our announcement in the past two Newsletters has been sufficient for us to soon announce scheduling for this fall.

We are encouraged to add two new workshops to the previous announcement:

- ☐ SOUND SYSTEM DESIGN WORKSHOP FOR CHURCHES & AUDITORIUMS
- ☐ FINANCIAL & MANAGEMENT SEMINAR/WORKSHOP

- ☐ STUDIO DESIGNERS WORKSHOP
- ☐ PZM™ & MICROPHONES FOR RECORDING & SOUND REINFORCEMENT WORKSHOP
- ☐ SOUND INSTALLERS SEMINAR/WORKSHOP
- ☐ INSTRUMENTATION WORKSHOP
- ☐ FUNDAMENTALS OF AUDIO

Don and Carolyn will host and assist in each of the special seminar/workshops.

We have printed this page of the Newsletter in looseleaf and included an extra copy so you can return it to us with your ideas. If you want, sign your name and we will contact you when the workshop/seminar you are interested in is planned. Or just mark your interests without your signature so that we have a feeling for your wishes.

NAME____

COMPANY______________________________________

ADDRESS______________________________________

CITY__________________STATE____________ZIP________

PHONE______________Month of year good for you:____________

If we haven't listed your special interest or a project leader that you would like to work with, please list here:______________________________________

____

HME - NEW SYN-AUD-CON SPONSOR

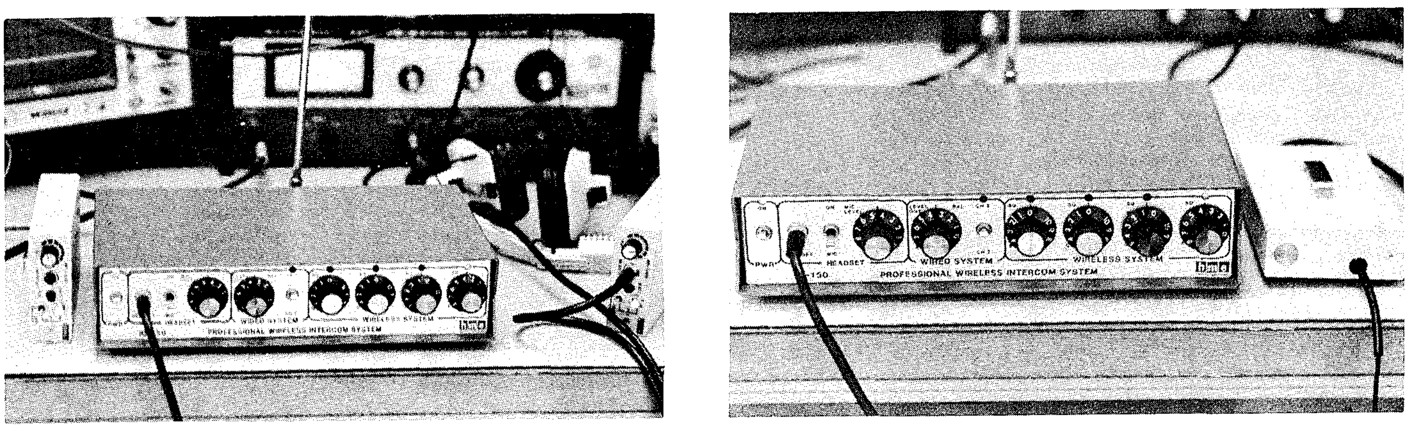
Syn-Aud-Con is pleased to announce a new sponsor--HME of San Diego, California.

The speciality of HME is wireless microphones. Syn-Aud-Con is very pleased to have as a sponsor a manufacturer of wireless microphones with the combined skill and integrity to supply Syn-Aud-Con graduates with a well-made, state-of-the-art system.

HME felt in such rapport with Syn-Aud-Con's educational goals that they expressed a willingness to undertake a sponsorship role with Syn-Aud-Con.

The president of HME is Harry Miyahira and we are pleased to discover that he has a thorough background in audio that includes having worked for our friend, Ralph Townsley at Purdue University. We know from experience that no one could be around Townsley without learning a great deal about the unwritten lore of audio. (Townsley is author of *Passive Network Design* published by Tab Books.)

In a recent visit to HME's extremely efficient manufacturing facility, we discovered HME's responsiveness to new ideas. Syn-Aud-Con is excited about several of the new ideas HME is working on and we expect to be reporting on them in the next Newsletter. We took a couple of pictures of the new professional quality wireless intercom system (battery operated) which will be in production soon.



HME Model IC150 Base Station, including integral four-channel mixer, interface model (for hard-wired systems), and head-set.

Harry is not only a fundamentally well-based audio and RF engineer but is developing a skilled management team which includes Robert Carr as Marketing Manager, who was with Shure Brothers in engineering and marketing for thirty years. Bob is a long time friend of Syn-Aud-Con and was instrumental in Shure becoming one of Syn-Aud-Con's earliest sponsors.

Wireless microphony has not yet solved all of its problems, such as RF interference, but Syn-Aud-Con has no hesitation in recommending HME as the source of honest, accurate advice on when (about 98% of the time) to use such devices with success and when to take other measures.

Syn-Aud-Con has used HME wireless microphones, without interference, all over the United States. We did have a source of interference at our Seminar Center when using the microphone at two classes this winter but *the source of the problem was a faulty shield connection in our system* which, upon being corrected, completely cleared up the interference. We use a PZM™ with our HME wireless microphones and have found them to provide an electrical signal at the output of the receiver identical to that obtained from a standard microphone cable between the microphone and measuring point.

We are enclosing literature from HME with this Newsletter and a new sponsorship sheet giving details of who to contact at each sponsor. We are very proud and pleased with the synergetic relationship that exists between Syn-Aud-Con, their sponsors and our graduates.

JANUARY 1981 CLASS

NEW DISCOVERY OF SABINE CORRESPONDENCE

The 1980's have begun with a series of interesting inquiries into the origins of architectural acoustics. John W. Kopec of IIT Research Institute, Riverbank Acoustical Laboratory in Illinois, unexpectedly found an additional cache of unknown, hence unpublished, notes and papers by Wallace Clement Sabine.

To quote the abstract of the article discussing this historic find that appears in the January, 1981, JASA, pages 1-16, "Twenty-two correspondence files of Wallace C. Sabine, discovered recently at the Riverbank Laboratory, contain a rich supply of solutions to acoustics and noise control problems." "Differing from his published papers ('Collected Papers on Acoustics,' edited by T. Lyman (Dover, New York 1964) and notebooks (J. Acoust. Soc. Am. 61, 629-639 (1977))), they are records of Sabine's extensive activities between 1909 and 1916 as an acoustical consultant."

There is correspondence reproduced in this article on auditorium acoustics, building noise transmission, and the reduction of noise from industrial plants and from machines. Sabine's role in the invention of acoustic tile and a view into his part in the development of Riverbank Laboratories is glimpsed here.

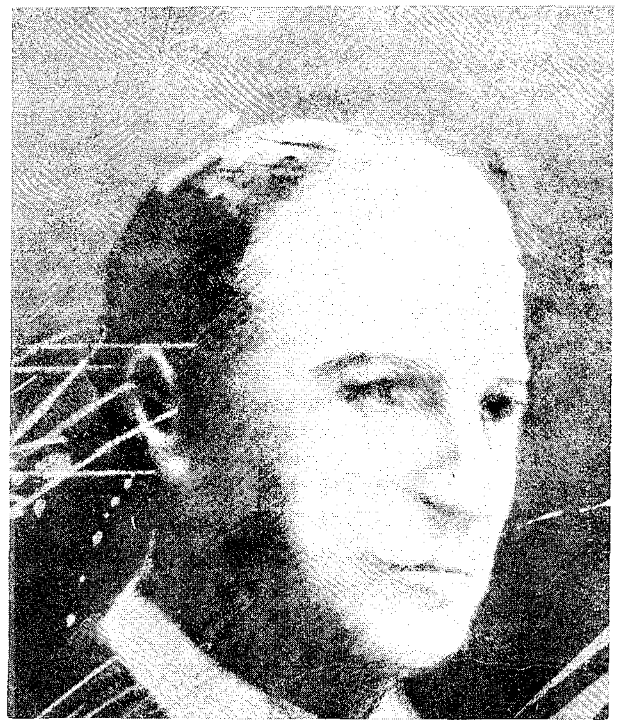
How many of us would like to qualify for this description of Sabine--"Known as a man of high principle and great modesty, possessing a far-ranging store of accurate knowledge, skilled at expressing himself and willing to extend help to anyone asking his advice."

These new papers include a hitherto unknown sketch of Professor Sabine.

Again, an idea of Sabine's impact on his clients is revealed in a letter to him dated September 20, 1911.

> Dear Mr. Sabine:
>
> You may have original drawings, photographs, furniture, good will, money, personal services, or any other old thing you want to ask for from this office, as we shall seize upon the opportunity with avidity. You do not seem to realize that you have put us everlastingly in your debt and that nothing we ever can do will be adequate compensation therefore.
>
> Very faithfully yours,
>
> Cram, Goodhue, and Ferguson

Finally, among this newly discovered correspondence is Sabine's classic answer to an architect's grumbling that "acoustics sometimes interfere with a building's appearance." Sabine's answer discloses a kind restraint in the use of a scapel-like brain.



Professor Wallace C. Sabine

> "In order not to feel myself an outcast in society, I have to stop every now and then and remind myself that whatever I may have done in architectural acoustics, I did not invent the problem. Please exonerate me from being the cause of your troubles. Hoping that nothing will be done, either in the correction of architectural acoustics <u>or in initial design</u>, which will interfere with architectural effects, I am......"

We can't resist one last quote from a newspaper clipping written by Boston's leading and respected music critic, William Foster Apthorp, in December, 1902, regarding the then brand new Boston Symphony Hall.

> "To begin with, neither the late Dr. Upham nor Mr. Sabine can be rightly deemed competent to express a musical opinion of any weight whatever; both come musically in the amateur class. And, to conclude with, we have not yet met the musician who did not call Symphony Hall a bad hall for music. Expert condemnation of the hall differs, as far as we have been able to discover, only in degrees of violence."

How familiar a sound "experts" have; how wrong they usually are; and how confident they are of their grasp of "ultimate truth." Like presidential assassins, they are primarily remembered for their ineptitude at any worthy task. Eighty years later Boston Symphony Hall is universally judged as the world's greatest concert hall.

We were privileged to hear Mr. Kopec's original presentation of this new material at last fall's ASA meeting in Los Angeles. We're grateful that the rediscovery of this material falls into such orderly intelligent hands and that once again Leo L. Beranek, as co-author of the article, draws upon his lifetime of dedicated research into Sabine's life and work.

USING FIXED ATTENUATOR "PADS" IN AUDIO "LINK" CIRCUITS

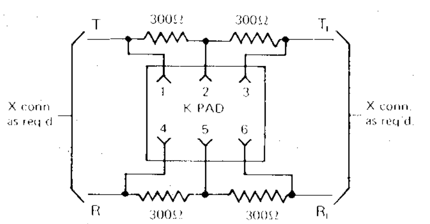
As is pointed out in every Syn-Aud-Con class during the discussion of "gain overlap" in electronic circuits, the use of "pads" is mandatory for *professional* results. Several key factors must be satisfied when "pads" are inserted in interconnecting "link" circuits between components in an audio system.

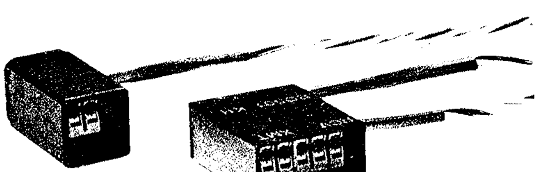
1. The output of the preceding component must be "built out" to the input value of the pad.
2. The following component must be "shunted" with the proper value to provide the pad its specified termination value.
3. The pad components should be properly housed and their attenuation value marked on the housing.

Syn-Aud-Con feels that this last point is very professionally met by using the Plantronics-Kentrox K-Pads and their associated easily mounted and wired housings. The mountings contain the necessary matching resistors and the "plug in" K-Pads contain the bridging and shunt resistors--a really slick package.

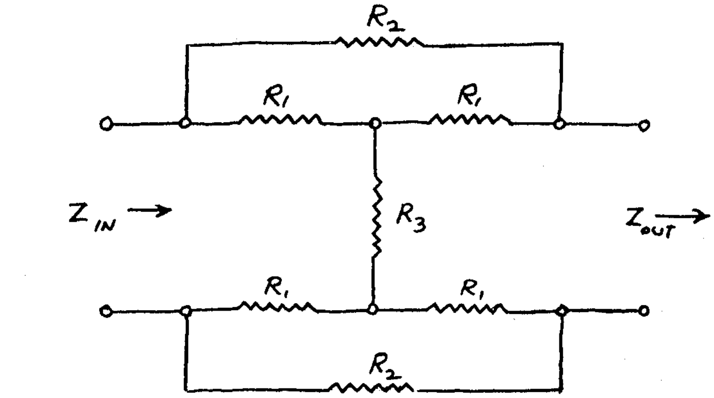
The data below tell their story. K-Pads not only work well but one glance into the rear of a rack instantly tells any *engineer* that someone really on the ball handled the design of that rack.

SIMPLIFIED SCHEMATIC DIAGRAM





CALCULATING BALANCED "BRIDGED 'H' PADS"



$R_1 = .5Z$, $R_2 = .5Z(K-1)$, $R_3 = Z\left(\frac{1}{K-1}\right)$

$K = 10\ EXP\ (dB\ ATTENUATION\ DESIRED/20)$, $Z_{IN} = Z_{OUT}$

EXAMPLE

A 600 Ω BRIDGED 'H' PAD WITH 6dB ATTENUATION

$K = 10EXP(6/20) = 2.0$, $R_1 = 600/2 = 300\Omega$, $R_2 = 300(2-1) = 300\Omega$

$R_3 = 600\left(\frac{1}{2-1}\right) = 600\ \Omega$

K-PAD® MOUNTINGS
Single and Double

Designed for 2 Wire (30000106) or 4 Wire (30000206) applications. Mounted with peel-off adhesive velcro (both sides supplied). Can be attached to most surfaces. Accept Kentrox K-Pad® plug-in attenuators, which provide fast, flexible, highly concentrated level-setting for voice and data applications. The K-Pad® attenuator circuit is electrically equivalent to the WECO 1C pad and 89-type resistors. K-Pads® are available in values from 0 to 35 dB in convenient increments. Used at customer premise for voice frequency telecommunication and data circuits.

Fortunately, K-Pads are sold in high volume to Telco users with the result that this highly professional equipment does not have an unreasonable pricing structure. We suggest you write for a copy of their catalog.

Plantronics
Kentrox Industries, Inc.
14335 N.W. Science Park Drive
Portland, Oregon 97229
(503) 643-1681

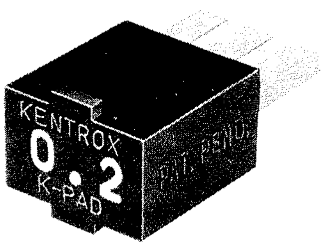
Continued on next page....

K-PAD SYSTEM

The patented K-Pad attenuator system (bridged H-Pads) provide concentrated level setting for voice and data circuits. It is electrically equivalent to WECO 1C Pad and 89 type resistors. This system is available for all commonly used mountings including 211, 183, 89A1-100, 89B1-100, NE-66QC-100, CCB, ESS 78A1-64 and 78B1-64, and others. 600 ohms is standard, but 900 ohms can be special ordered.

K-Pad attenuator modules These modules provide attenuation values from 0.0 dB to 35.0 dB in increments as small as 0.2 dB. They're easy to install and remove and allow high concentrations per mounting.

The model numbers listed below are for 600 ohm mountings.



Attenuator in dB	Part Number	Bell System Common Language Code	Attenuator in dB	Part Number	Bell System Common Language Code	Attenuation in dB	Part Number	Bell System Common Language Code
0.0	40000	PDOK0006AA	8.4	40840	PDOK0846AA	16.8	41680	PDOK1686AA
0.2	40020	PDOK0026AA	8.6	40860	PDOK0866AA	17.0	41700	PDOK1706AA
0.4	40040	PDOK0046AA	8.8	40880	PDOK0886AA	17.2	41720	PDOK1726AA
0.6	40060	PDOK0066AA	9.0	40900	PDOK0906AA	17.4	41740	PDOK1746AA
0.8	40080	PDOK0086AA	9.2	40920	PDOK0926AA	17.6	41760	PDOK1766AA
1.0	40100	PDOK0106AA	9.4	40940	PDOK0946AA	17.8	41780	PDOK1786AA
1.2	40120	PDOK0126AA	9.6	40960	PDOK0966AA	18.0	41800	PDOK1806AA
1.4	40140	PDOK0146AA	9.8	40980	PDOK0986AA	18.2	41820	PDOK1826AA
1.6	40160	PDOK0166AA	10.0	41000	PDOK1006AA	18.4	41840	PDOK1846AA
1.8	40180	PDOK0186AA	10.2	41020	PDOK1026AA	18.6	41860	PDOK1866AA
2.0	40200	PDOK0206AA	10.4	41040	PDOK1046AA	18.8	41880	PDOK1886AA
2.2	40220	PDOK0226AA	10.6	41060	PDOK1066AA	19.0	41900	PDOK1906AA
2.4	40240	PDOK0246AA	10.8	41080	PDOK1086AA	19.2	41920	PDOK1926AA
2.6	40260	PDOK0266AA	11.0	41100	PDOK1106AA	19.4	41940	PDOK1946AA
2.8	40280	PDOK0286AA	11.2	41120	PDOK1126AA	19.6	41960	PDOK1966AA
3.0	40300	PDOK0306AA	11.4	41140	PDOK1146AA	19.8	41980	PDOK1986AA
3.2	40320	PDOK0326AA	11.6	41160	PDOK1166AA	20.0	42000	PDOK2006AA
3.4	40340	PDOK0346AA	11.8	41180	PDOK1186AA	20.2	42020	PDOK2026AA
3.6	40360	PDOK0366AA	12.0	41200	PDOK1206AA	20.4	42040	PDOK2046AA
3.8	40380	PDOK0386AA	12.2	41220	PDOK1226AA	20.6	42060	PDOK2066AA
4.0	40400	PDOK0406AA	12.4	41240	PDOK1246AA	20.8	42080	PDOK2086AA
4.2	40420	PDOK0426AA	12.6	41260	PDOK1266AA	21.0	42100	PDOK2106AA
4.4	40440	PDOK0446AA	12.8	41280	PDOK1286AA	21.2	42120	PDOK2126AA
4.6	40460	PDOK0466AA	13.0	41300	PDOK1306AA	21.4	42140	PDOK2146AA
4.8	40480	PDOK0486AA	13.2	41320	PDOK1326AA	21.6	42160	PDOK2166AA
5.0	40500	PDOK0506AA	13.4	41340	PDOK1346AA	21.8	42180	PDOK2186AA
5.2	40520	PDOK0526AA	13.6	41360	PDOK1366AA	22.0	42200	PDOK2206AA
5.4	40540	PDOK0546AA	13.8	41380	PDOK1386AA	22.2	42220	PDOK2226AA
5.6	40560	PDOK0566AA	14.0	41400	PDOK1406AA	22.4	42240	PDOK2246AA
5.8	40580	PDOK0586AA	14.2	41420	PDOK1426AA	22.6	42260	PDOK2266AA
6.0	40600	PDOK0606AA	14.4	41440	PDOK1446AA	22.8	42280	PDOK2286AA
6.2	40620	PDOK0626AA	14.6	41460	PDOK1466AA	23.0	42300	PDOK2306AA
6.4	40640	PDOK0646AA	14.8	41480	PDOK1486AA	24.0	42400	PDOK2406AA
6.6	40660	PDOK0666AA	15.0	41500	PDOK1506AA	25.0	42500	PDOK2506AA
6.8	40680	PDOK0686AA	15.2	41520	PDOK1526AA	26.0	42600	PDOK2606AA
7.0	40700	PDOK0706AA	15.4	41540	PDOK1546AA	27.0	42700	PDOK2706AA
7.2	40720	PDOK0726AA	15.6	41560	PDOK1566AA	28.0	42800	PDOK2806AA
7.4	40740	PDOK0746AA	15.8	41580	PDOK1586AA	29.0	42900	PDOK2906AA
7.6	40760	PDOK0766AA	16.0	41600	PDOK1606AA	30.0	43000	PDOK3006AA
7.8	40780	PDOK0786AA	16.2	41620	PDOK1626AA	35.0	43500	PDOK3506AA
8.0	40800	PDOK0806AA	16.4	41640	PDOK1646AA			
8.2	40820	PDOK0826AA	16.6	41660	PDOK1666AA			

"DO WE HEAR DAMAGING NOISE CORRECTLY ?"

Those attending Syn-Aud-Con classes know our response to government noise regulations. We are aware that sincere, competent workers are occasionally attracted to this type of work but we also are all too aware of the political mindedness of many who see the entire scene as an invitation to set up a financial security system for "noise control" experts.

The article by P. V. Brüel included in this mailing through the courtesy of Brüel & Kjaer is, in our opinion, the most sensible, believable, and useful data on this subject to have come into our hands. Figure #9 on the 6th page is a truly marvelous aid in visualizing some of the temporal attributes of human hearing. Syn-Aud-Con's experience strongly supports Mr. Brüel's conclusions and we feel the authorities will continue to chase their tails until they heed what he has to say.

I hope you will read the article carefully.

BILL PUTNAM OF UNITED RECORDING

Syn-Aud-Con found the "Behind the Scenes" article in *Audio* magazine written by Bert Whyte of such interest that we felt our readers would enjoy it as well.

Syn-Aud-Con classes currently use 813 Time Align™ monitors, the new UREI 6500 self-correcting amplifier, the UREI 200 level recorder and, of course, UREI equalizers.

Bert Whyte's article is about the man behind all of these fascinating and innovative products. Perhaps Bill Putnam's highest accolade is the caliber of men who work for him at UREI. Syn-Aud-Con has personal experience with the consistent creativity of these men as well as their extremely likeable human attitudes. Bill Putnam's interaction with these men establishes in our mind his ability to judge technological talent accurately and the even rarer ability to allow its growth within liveable business constraints. With these worthwhile traits in mind, we believe you'll enjoy Bert Whyte's article on Bill Putnam and his contribution to our industry.

Most of us are familiar with the founding fathers of high fidelity. Pioneers like Avery Fisher, Rudy Bozak, Frank McIntosh, H.H. Scott, Walter Stanton and others, whose dedication to the quest for high-quality sound reproduction resulted in their manufacturing the first hi-fi audio components for the consumer market.

Needless to say, this fledgling activity in the consumer hi-fi market was complemented by parallel developments in the fields of professional audio and recording. There were pioneers in both these fields as well, but within the relatively circumscribed world of professional audio, few people achieved high visibility and their accomplishments have largely gone unrecognized.

One of the most gifted of those early pioneers in professional audio is Milton T. (Bill) Putnam, currently Chairman of the Board of United Recording Corp. in Hollywood, California. URC encompasses United/Western Studios, Coast Recorders, Teletronix, and its manufacturing arm UREI (United Recording Electronics Industries).

Bill Putnam founded Universal Recorders and built his own studios in Chicago in 1946. It must be remembered that in those days, most recording studios were constructed with only rudimentary knowledge and little application of acoustic treatment. Typical studios of 15,000 to 35,000 cubic feet used draping and perforated acoustic panels, quite often applied directly to boundary surfaces with no provision for air space behind them. Some used rock-wool batts behind perforated Celotex. For the most part, there was inadequate low-frequency absorption with the ratio of indirect-to-direct sound in the low-frequency instruments causing a distinct lack of separation and presence and an unpleasant coloration of the sound. Add to this the fact that the off-axis response of many microphones (even bidirectional or cardioid types) caused time-related and spectral colorations of a signal arriving after the direct sound. Of course, at the low frequencies below 125 Hz, all the directional mikes became virtually omnidirectional, further contributing to the muddiness of the sound.

In marked contrast to this common studio environment, Bill Putnam's studio at Universal Recorders was constructed with the specific goal of increasing instrumental separation, and he accomplished this by lowering the overall reverberation time, with particular attention to substantial absorption of the low frequencies. To this end, diaphragmatic panels in convex splays were used, and the thickness of rock-wool batts behind perforated panels was increased by furring out frames for greater depth. Separation screens and rugs were used for absorption of higher frequency reflections.

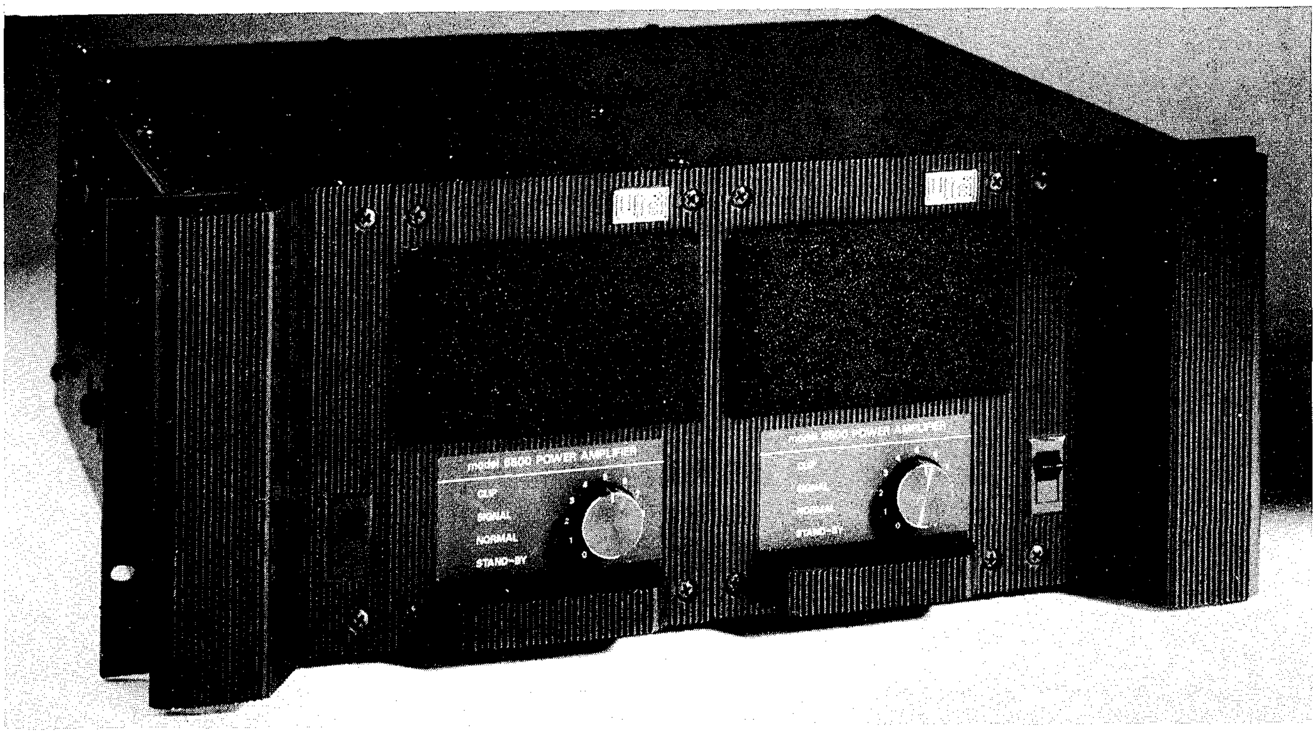
An early type of absorptive roll-around isolation vocal booth was used, which also was occasionally used for a drum set — a forerunner of today's drum cage. A roll-around band shell was constructed with interior polycylindrical diffusers to prevent focusing, especially of strings. Instrumental positioning and mike placement in the studio were radically altered from the hand-me-down practices of early broadcasting and recording, with a view towards more separation and definition. (Remember, this was before the days of multi-track and overdubbing.) Having achieved better instrumental separation by lowering studio reverb time, Bill did not want a dead or dry overall sound, so he experimented with feeding his output signal to various types of reverberation rooms. (Note that this was before electronic echo chambers.) Bill was looking for a reverberation room which had a smooth decay, would eliminate periodicity, and have low coloration. Bill experimented with delaying the signal sent to the reverb room to more effectively simulate early sound. He realized the importance of the control room in his studio, and it was acoustically treated. This led to measuring monitor-speaker response in the room and even some attempts at room/speaker equalization.

Because of all the factors in studio design and recording practices Bill had incorporated into his Universal Recorders studios, he developed a distinctive and easily recognized high-quality sound which generated considerable business. In the early and mid-fifties, the advent of exotic high-quality condenser microphones such as the Telefunken U-47, along with greatly improved mono feedback cutterheads such as the Westrex 2B and Grampian, resulted in rapid advances in the quality of recordings. Bill Putnam adopted these and other aids to high-quality mastering, and in competition with several other studios, the "Hi-Fi Spectacular" record was a new specialty product for what was called the audiophile market! Imagine that, way back in 1955! As tape recorders and tape formulations improved, permitting greatly extended high-frequency response, the battle to produce even more spectacular hi-fi records intensified. In 1955, Bill managed some neat technical one-upmanship by recording at 30 inches per second and mastering the tapes at half speed. Shades of Mobile Fidelity back then!

I had the pleasure of meeting Bill Putnam in 1951 in Chicago, where I was in technical sales and served as music director for Magnecord, one of the pioneer manufacturers of tape recorders. This is where stereophonic sound enters the picture. I had been recording in stereo for some time, and Bill had been experimenting with stereo too. The Magnecord stereo recorder used a staggered head configuration which gave adequate stereo separation but also caused certain problems. Bill recorded some stereo in this format. Ever the innovator, in 1954, in cooperation with the Pentron Corp. of Chicago, he used a recorder with eight channels on standard quarter-inch tape in a staggered head configuration to record an instrumental group. In the eight-channel playback, eight loudspeakers were arrayed in the listening room in the same positions in which the instruments had originally been recorded. Portents of a future as yet unrealized.

In 1958, Bill Putnam moved to Hollywood, California, formed United Recorders and constructed the first purpose-built studio for stereophonic recording. The studio was an amalgam of Bill's original ideas, with important new acoustic design work by Michael Rettinger. In later revisions, the Time Delay Spectrometry (TDS) measurement system of Au-

Continued on next page.....



dio's own Richard Heyser was extensively employed, as were some of Don Davis' pioneering ideas on room equalization. John Eargle contributed new data on monitor speaker response in acoustically treated control rooms. Bill Putnam and Ed Long (a frequent contributor to *Audio)* collaborated on a joint R&D program to upgrade the quality of monitor speakers. The result was a speaker which was designated a "Time Aligned" studio monitor. The technique takes into consideration the time (phase) response of the speaker, and by proprietary design of the crossover network and the placement of the loudspeaker driver elements, the system is said to be "Time Aligned" and free of group time-delay anomalies. In essence, there is near-perfect alignment of the frequency components in a complex transient waveform.

Bill Putnam had formed a company — United Recording Electronics Industries, UREI — to manufacture specialized equipment for the professional recording market, and the new "Time Aligned" studio monitor was to be built at this facility. UREI has been in existence for some years now and is one of the most successful of Bill's companies. I have used a number of UREI's more exotic products, and when I was invited to visit their new plant, I gladly accepted.

The UREI plant is a modern one-story building in Sun Valley, one of those pleasant little towns in the San Fernando Valley, and just a short drive to the United Recording studios in Hollywood. DeWitt "Bud" Morris, President of UREI, is an old friend. He's an easygoing and genial man, but nonetheless runs a tight ship at UREI, aided by Brad Plunkett, Chief Engineer, and Ray Combs, V.P. Work flow is very well organized, beginning with in-house testing of incoming parts, then sub-assembly build-up and unit fabrication on various dedicated technology assembly lines. While vendors supply many parts, whenever possible and economically feasible, UREI makes their own. For example, most transformers and certain coils are made on the premises. UREI is subject to the same inordinately long delivery times on parts as most companies these days, but they invest heavily in quantity buying and as a consequence are usually in a strong inventory position. Thus, most of the equipment they make is readily available from stock.

In addition to a well-equipped R&D lab, a modern test lab monitors quality-control procedures. Fortunately, subjective testing of many UREI products is given high priority and is carried out in a purpose-built, acoustically treated listening room. Currently UREI is producing a new generation of "Time Aligned" monitor speakers, the 811A (single 15-in. woofer with exponential HF horn), the 813A (same as 811A with addition of another 15-in. woofer in a large enclosure), and 815A (same as 811A with two 15-in. woofers in a still larger enclosure). The HF horns are newly designed for improved frequency response and dispersion characteristics. The "Time Aligned" crossover network has been redesigned for more power transfer to the speaker driver.

UREI is now heavily committed to amplifier production. Their big brute, Model 6500, was introduced at the May, 1980 AES Convention, and I described it in the August, 1980 issue of *Audio*. At the November, 1980 New York AES, the 6500 was joined by the 255 W/channel Model 6400, the 150 W/channel Model 6250, and a slim-line design, only 1 ¾-in. high, the Model 6150 with 76 W/channel. All of these units are 19-in. rack-mountable and are of highly rugged construction.

UREI continues to produce such specialized items for professional recording as limiters and compressors and various types of equalizers and filters. A new electronic crossover is currently in the works, and UREI still makes their invaluable Model 200 XY plotter and recorder with the Model 2000 frequency response module, the Model 2010 level and frequency detector module, and the new Model 21 warble generator for room measurements. When I was visiting the UREI plant, a huge new Neve mixing console with the Necam automated mixdown feature was undergoing tests prior to installation in Bill's studio at United Recorders.

Over the years, Bill Putnam has always tried to stay just a bit ahead of current recording technology. His early experiments with time delay and subsequent manufacture of the Cooper Time Cube is an example, as was his issuing of four-channel matrix evaluation test records. Bill's quadraphonic recordings of the late Stan Kenton and his orchestra are among the very best ever done in this medium. Today Bill Putnam continues in his pursuit of recording excellence, and his UREI company translates many of his ideas into products that find favor in professional audio.

WOMEN IN AUDIO

Letter from NYYA LARK, Recording and Live Sound Engineer at Fantasy Records in Berkeley, California, to any and all interested in supporting an organization called *Women in Audio*:

"Hello Again,

"It's definitely been quite awhile since last spring's Audio Engineering Society's convention, when we came together for the first Women In Audio Session.

"Those of us on the panel felt that such a momentous occasion should not be left to history alone or just fond memories, but should be continued. Especially now that we are fully aware of the good amount of women in the audio industry and have a working idea of how many would be interested in becoming involved.

"Hence, the idea of a newsletter. We thought it would definitely be a good way to stay in touch with the women in the industry and find out what they're doing, share mutual experiences, new concepts, problems and all practical and scientific endeavors in the way of the female aspect. It would also be an excellent vehicle to help those get started who had no idea how, and to inform the non-professional as well as the newcomers what we are about.

"I certainly hope that this idea becomes a reality. However, as with many things in life, money is the key issue besides the most needed communication on all of our parts. If each of us contributed $10, that would be enough to get the presses rolling, so to speak. After that, yearly dues would be set and naturally bigger and better issues and not to mention, your donation is tax deductible!!

"So please help by sending your contribution of money, ideas, thoughts and experiences or anything that you feel would be beneficial to others and yourself. You can also pass this thought along to others who may be interested. And no, it's not a woman libbers move either!!! I know that there are interested male parties that you may know out there too!

"If you're on the East Coast, please contact:

Mary Gruszka
CBS Television Network
51 West 52nd Street
New York, New York 10019

"If you're on the West Coast, you can contact:

Nyya F. Lark
3933 Harrison St. #201
Oakland, California 94611

"Truly hoping to hear from you soon.

"Nyya F. Lark"

A *Women in Audio* organizational meeting will be held in the Syn-Aud-Con hospitality suite at AES in May. As Nyya says, "I certainly hope that this idea becomes a reality."

If you were not able to attend the May AES and would like to support the organization, be in touch with Nyya Lark or Mary Gruszka.

METERS AND VOLTMETERS

Meters measure *flow*.

EXAMPLES

Water Meter

Watt Meter

Voltmeters, etc., that do not measure *flow* are instruments.

TIME-BANDWIDTH PRODUCT OF FFT ANALYZERS

Number of lines (resolution)	(L)	$L = t\,(B.W.)$
Time window in secs.	(t)	$t = \frac{(B.W.)}{L}$
Bandwidth in Hz	(B.W.)	$B.W. = \frac{t}{L}$

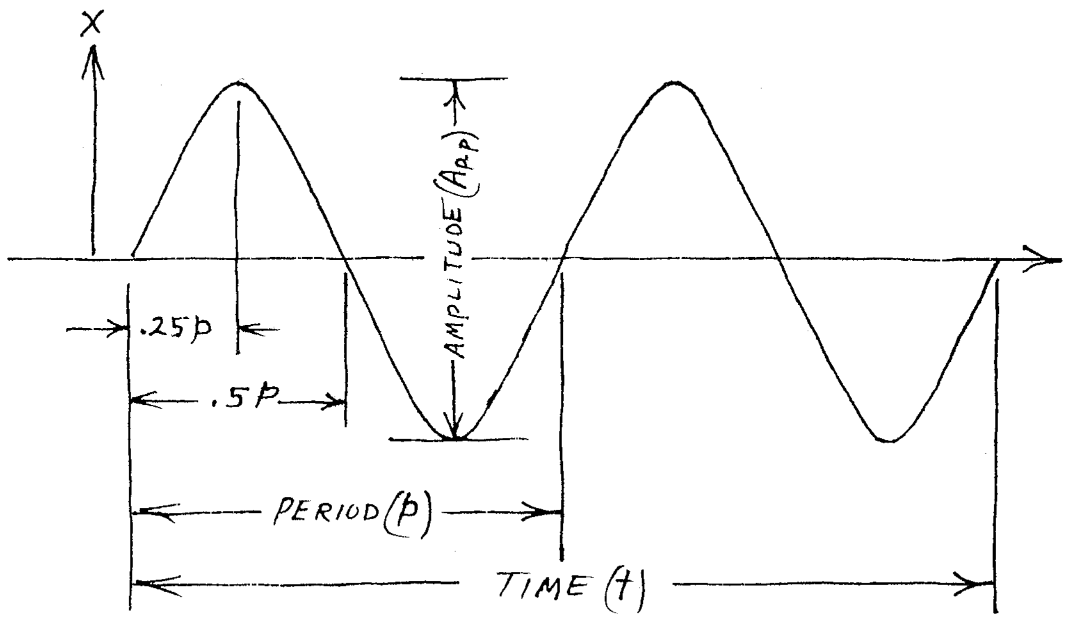
EXAMPLE

An FFT analyzer with 400 line resolution and a bandwidth of 500 Hz has a time window of:

$$t = \frac{(B.W.)}{L} = \frac{500}{400} = 0.8 \text{ sec.}$$

FEBRUARY 1981 CLASS

THE SIMPLE SINE WAVE



A simple sine wave (sin x) has the following basic parameters:

1. A period (p) — The primitive period (p) of sin x is 2π or 360° - i.e. one cycle.
2. An amplitude measured peak-to-peak (Ap-p) — The amplitude may be in volts (V), current (I), etc.
3. A time (t) — The time interval is Np in secs.

From these basic parameters we derive:

1. Phase (P) — $P = \left(\frac{t}{p}\right)$ and is a ratio
2. Phase angle (Θ) — $\Theta = 360\left(\frac{t}{p}\right)$ in degrees $\Theta = 2\pi\left(\frac{t}{p}\right)$ in radians
3. Frequency (f) — $f = \left(\frac{p}{t}\right)$ in Hz
4. Wave length (λ) — The wave length is the distance between points of corresponding phase of two consecutive cycles.

$$\lambda = \frac{\text{Phase Velocity}}{f}$$

5. $.5A_{p-p}$ = Peak amplitude (Ap)
   $.5Ap$ = Average amplitude (A_{av})
   $.707Ap$ = Root mean square amplitude (A_{rms})

   — For sine waves only

HAVING FUN WITH CALCULATORS

$$10 \log_{10} 2 = \log_{\sqrt[10]{10}} 2 = 3.01$$

Because:

$$a/c = b^N$$

$$\ln a/c = \ln b(N)$$

$$\ln b = \frac{\ln a/c}{N}$$

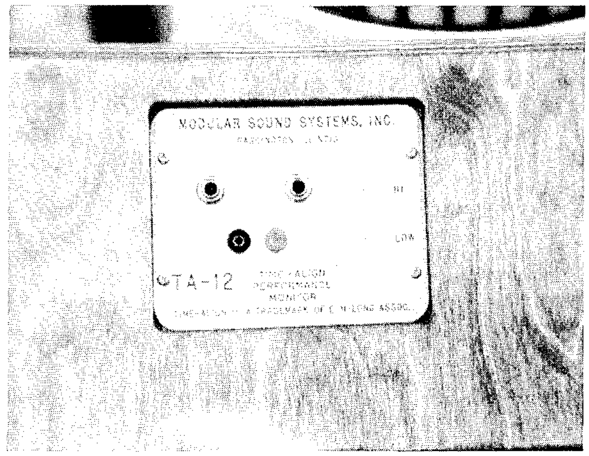
$$b = e^{\left(\frac{\ln a/c}{N}\right)}$$

Therefore: $b = e^{\left(\frac{\ln 2}{3.01}\right)} = \sqrt[10]{10} = 1.26\cdots$

Thus: $(\sqrt[10]{10})^{3.01} = 2$

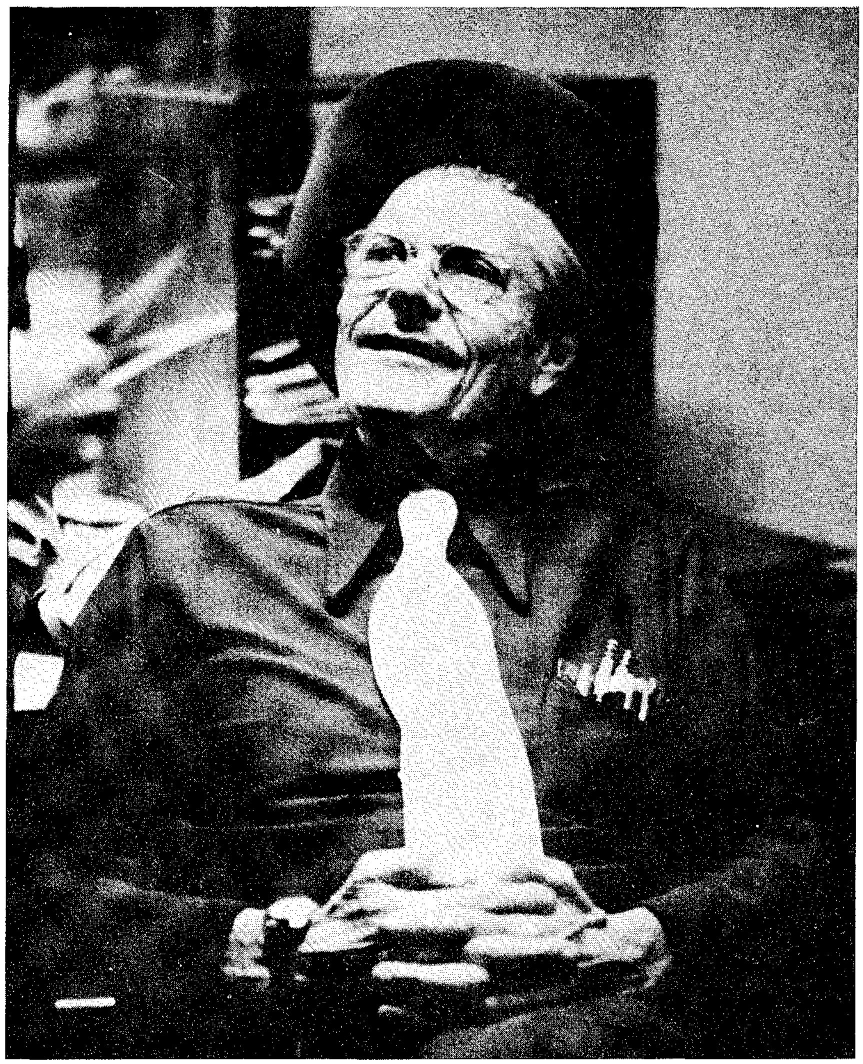
BAG END TIME-ALIGNED™ MONITOR

JIM WISCHMEYER of Bag End, Barrington, Illinois, builds rugged stage monitors. He recently came up to show us his latest version which he has had made into a *"Time-Align™" Performance Monitor* by E. M. Long Associates.



We were able to verify the success of Ed Long's work in achieving alignment on axis, and to a surprisingly large off-axis angle, as well. We were particularly pleased that Jim thought enough of his product to spend the time and money to perfect this aspect of its performance.

PAUL KLIPSCH



Just in case you were so unfortunate to overlook it, there was an article on Paul Klipsch in the August, 1980, issue of "Audio" magazine. The article includes some recent photographs and the usual fascinating interview with the great man.

Writing about Paul Klipsch has many of the same hazards as writing about General Patton. Both have an impish quality that revels in misdirecting the interviewer away from their remarkable intellectual accomplishments and towards their outrageous contributions to trivia. Paul once again succeeds in so doing. The interview is entertaining, but not enlightening.

Paul is most enlightening when cornered by a question on which he has to start from scratch. You then witness the raw power of a great mind trapped into unaccustomed challenge, and the angles from which he attacks the problem would, you quickly find, never have occurred to you in a million years.

An American original?--You bet--but not in humor--rather in his ability to inspire and motivate late starters into imitating his persistence and continuing study.

DON'T SPLIT SPEAKERS

The object of a sound reinforcement system is a simple one. It is to increase the acoustic power in a given environment in a manner that insures that each listener gets an equal share and that as little as possible goes to areas where there are no listeners.

If we view this goal in terms of how to most effectively apply loudspeakers in achieving it, a number of preferred practices are available and deviations from these practices lead directly to inferior performance.

Whenever possible, use only *one* loudspeaker correctly placed. Correctly placed is directly above the talker at a distance that allows sufficient acoustic gain to be developed.

MAJOR COMPROMISE #1

When one loudspeaker cannot evenly cover all the listeners with equal acoustic power, good practice dictates two loudspeakers *at the same location*--one (more directional) aimed at the farthest listeners and another (less directional) aimed at the nearest listeners.

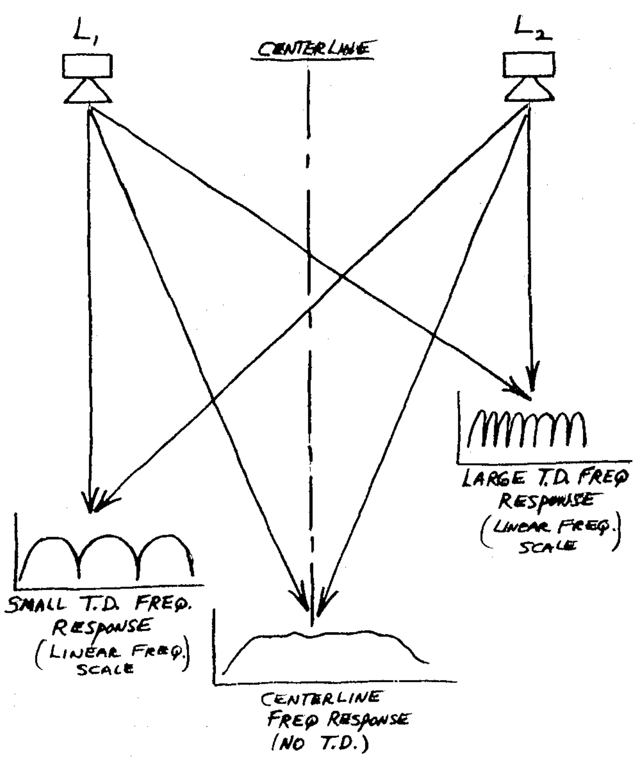
MAJOR COMPROMISE #2

When long distances are encountered and existing available loudspeakers cannot effectively "throw" sound that far, the preferred practice is to spread them out overhead as a series of single source systems and then align them to approximate a single loudspeaker through the use of digital time delay devices.

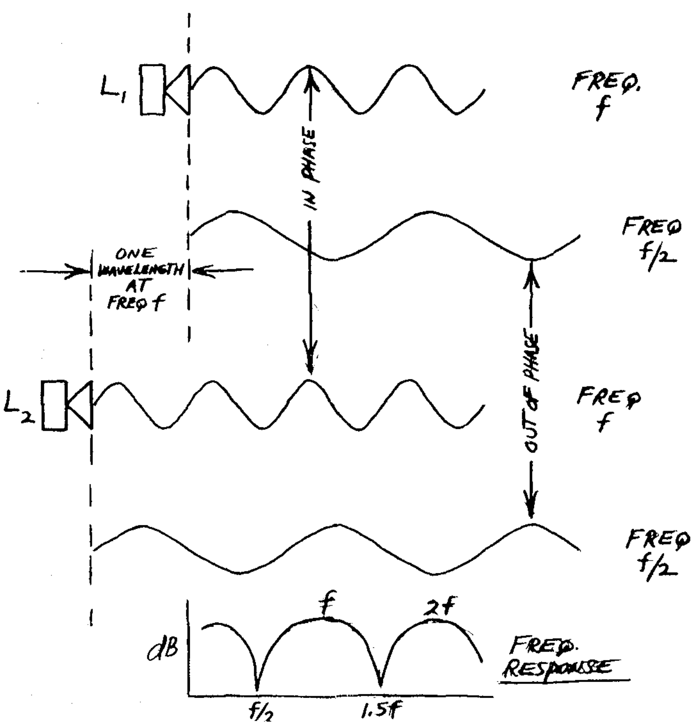
MAJOR COMPROMISE #3

When the first three approaches fail to meet the requirements of equal acoustic power to all listeners, then high density overhead distributed loudspeakers must be used. In cases of high reverberation or abnormally high ambient noise levels this may become, in the limiting case, a "pew back" loudspeaker system in a church wherein each two listeners is served by their own loudspeaker mounted in the backs of the pews.

RESPONSE OF "SPLIT" LOUDSPEAKER SYSTEMS



CREATION OF A "COMB FILTER" RESPONSE



WHY ARE THESE CONSIDERED GOOD?

A most reasonable question. The complete answer is multifaceted but several significant effects can be easily described.

The sole possibility of interference with a single loudspeaker system is that caused by reflections. Care in placement (high enough, angled down, etc.) and consideration of the matching of the narrowest coverage angles commensurate with delivering equal acoustic power to all listeners can greatly minimize such effects.

The instant more than one loudspeaker is used there arises the danger of two identical *equal power* signals arriving at a listener's ears at slightly different time intervals. This causes what is know as "comb filters."

Continued on next page.....

"Comb filters" are a frequency response full of peaks and nulls caused by "phase" cancellation--that is, a wavelength displacement in time that results in a frequency dependent series of anomolies in the frequency response curve. The easiest example at hand and one of the most common errors made in sound reinforcement work is the use of *two* loudspeakers on either side of a stage, platform, or other performing area. As can be seen in the illustration, the only "good" seats are in the "center aisle." Slight displacements to either side of center results in severe "comb filtering" of the frequency response. (We've all heard this many times--that funny hollow sound in our heads.)

Overhead distributed loudspeakers, when used in sufficient density, tend to have pairs that cause these anomolies and other pairs creating peaks where the first pair made nulls and vice versa.

Time delayed channels are normally put 20 msec (i.e., twenty feet) apart acoustically to utilize the Haas effect and thus produce very narrow tightly packed anomolies. Only the single source system avoids the initial cause (so long as a reflection is not allowed to act as an *equal power* vertical source).

These are your choices--some better, some worse. You now know what is considered to be "good practice" and why. It only takes skilled application of this knowledge to produce improved sound systems.

WORK STARTED ON THE SYN-AUD-CON PROGRAM LIBRARY FOR HP-41 USERS

At the Syn-Aud-Con/Heyser TEF™ Seminar in February at Syn-Aud-Con's West Coast Seminar Center, RUSSELL BERGER of Dallas, Texas, KEN WAHRENBROCK of Downey, California, JOHN LANPHERE of South Bend, Indiana, and FARREL BECKER of Washington, D.C., put their heads together and decided to begin establishing a library of programs for the users of HP-41 calculators.

They will act as a team to gather, edit and document programs contributed by Syn-Aud-Con graduates and others interested in Audio Design Programs.

John Lanphere will act as the program librarian and will coordinate the compilation of the library. Programs will be made available later this year. Bar Code listings and magnetic cards along with complete documentation will be available for each program at a nominal cost.

JOHN LANPHERE putting together library.

KEN WAHRENBROCK and RUSS BERGER working on programming

It is also possible that as a standard assortment of often-used programs is formed, these programs will be made available as a custom Syn-Aud-Con Applications Module.

We need to know who all of you HP-41 users are, so please write John Lanphere, A/V Design Service, 1408 Elwood Ave, Ste 213, South Bend, IN 46628, 219-234-1991. Be sure to send a self-addressed stamped envelope.

Be sure to tell him about any original programs that you would like to contribute, any ideas for programs you would like to see written, and any other ideas or suggestions regarding the program library.

Contributions accepted as part of the Syn-Aud-Con library entitle the contributing author to three (3) free selections among other available programs in the collection.

Syn-Aud-Con hopes that this effort to coordinate the remarkable diversity of programming talent among our graduates will meet with complete success.

WHO WILL BE THE FIRST TO TRY
A RADICAL SUGGESTION FOR IMPROVED MONITORING?

EXCERPT FROM
NEWSLETTER
VOLUME 7 # 3

Having now played at length with a high frequency "sled test" on the Emilar system, we have a number of subjective impressions.

1. You sure can hear a few inches of misalignment.

2. Woofers driven with a sine wave signal at crossover don't sound like the high frequency unit driven at crossover with the same signal.

We know that at least part of the difference may lie in the difference in both Q and C_ℓ at crossover between the woofer and tweeter.

In our experience many really good woofers have a Q = 2 or 3 while it is our considered opinion that high frequency units for studio monitoring work should have Q_S from 10 to 15.

There is also the problem of achieving really good bass response using the studio window wall as part of the surface for your low frequency development.

One solution to this dilemma is to mount the high frequency units about operator head level on either side of the window between the control room and studio and the low frequency unit overhead in the ceiling. Why?

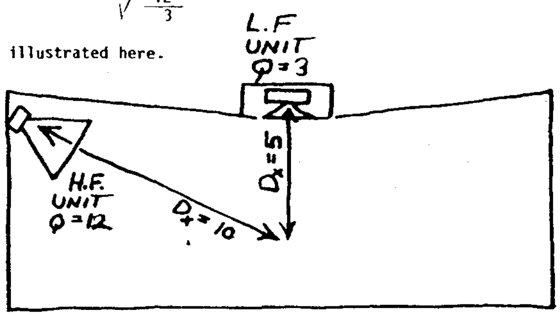
AN EXAMPLE

Suppose that our high frequency unit has a Q = 12 and our low frequency unit has a Q = 3. We find that D_2 from the H.F. unit to the operator's ears is 10 feet. What should the L.F. unit's D_2 distance be for an equal ratio of direct to reflected sound to be received at the operator's ears?

$$D_{LF} = \frac{10 \text{ feet}}{\sqrt{\frac{12}{3}}} = 5 \text{ feet}$$

A drawing of the installation is illustrated here.

$$D_{x\ L.F.} = \frac{D_{x\ H.F.}}{\sqrt{\frac{Q\ H.F.}{Q\ L.F.}}}$$



The imminent advent of digital time delays with adjustments in usec increments bodes well for the timeliness of this, up to now, radical idea. One usec is the equivalent resolution of .01356 inches.

EXCERPT FROM "db" Magazine, March 1981, page 43; article by Michael Rettinger, "Reproducing Electronic Music in the Control Room"

A horn with a length of about 8.3 feet (99.6 inches) would probably have a total length of 10 feet, once the woofer is installed at the throat. To shorten this dimension, a folded horn may be used. FIGURE 3 shows such a unit, as was employed for the reproduction of those very low notes in Universal Studios' "Sensurround" system. The system is intended to reproduce not only these low-frequency sound waves, but also wind rumbles and other below-25 Hz effects.

FIGURE 4 shows the floor plan and elevation of a recording studio control room designed to accomodate the horn system just described. To facilitate construction, the horn mouths may be either square or rectangular. Because of the frequencies that are reproduced, these units are practically omni-directional, and need not be installed in the front wall of the room.

If digital recording technology eventually allows us to routinely record very-low frequencies, perhaps the audiophile listening room of the future will also see the installation of such horn systems, when (and if) these sounds make their way to the lp record. ■

References

The best sources of information about loudspeakers and sound reproduction are the books by H. F. Olson. His first book "Applied Acoustics" has become a rarity because it is out of print. It was originally published by P. Blakiston's Son & Co. Inc., Philadelphia. It was written in collaboration with Frank Massa, a Swope Fellow at MIT. Two other books by H. F. Olson are "Acoustical Engineering" and "Modern Sound Reproduction," both published by Van Nostrand Co. Inc. in Princeton, New Jersey.

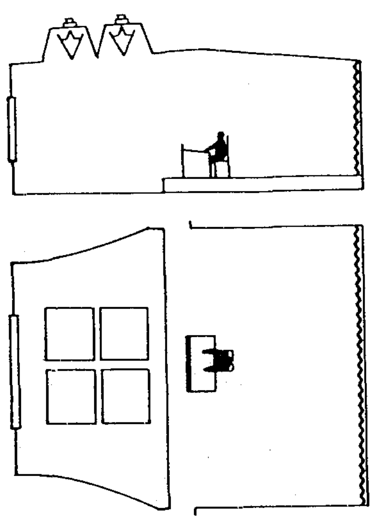


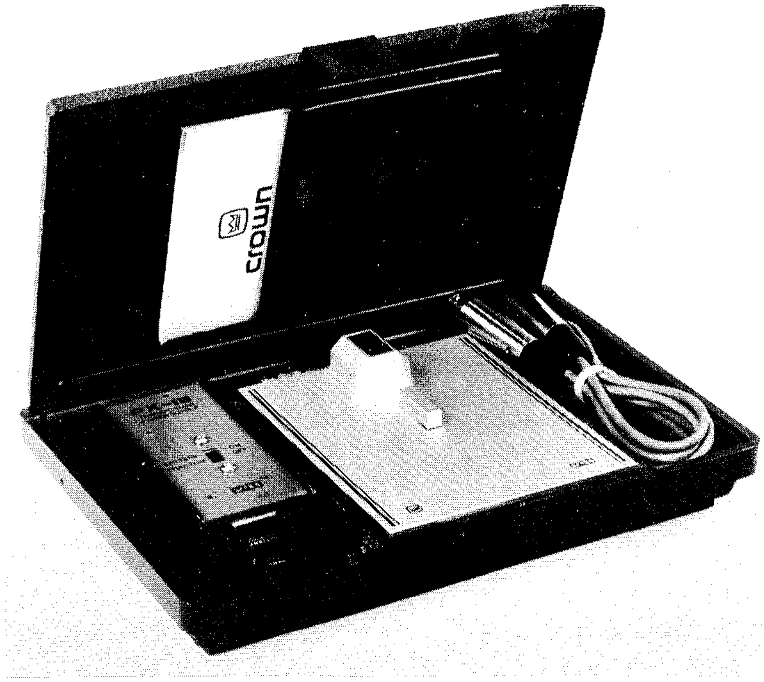
Figure 4. Low-frequency horns installed in Control Room ceiling.

"Who will be the first to try a radical suggestion for improved monitoring? Michael Rettinger?

PZM™

Syn-Aud-Con graduates will one day tell their grandchildren, "I was there when it all happened," when those grandchildren, after looking at one of today's conventional microphones in a museum, ask, "Grandpa, did you ever use one of those?" By then PZM™ will have solved a great majority of the world's input transducer problems and the strange monstrosities of today will look like what they really are--a crude approximation.

Syn-Aud-Con is truly impressed by the engineering advances and the striking cosmetic improvements Crown has built into their product.



Have you stopped to think about and realize that the purchase of the beautiful new Crown PZM™ packages not only provides you with the most advanced microphone capabilities in the world today but, if cared for and preserved, will remain a prize collector's item in the future?

We can't help but remember that only eleven years later the HP-35 calculator that we put aside (the only one we saved out of over thirty we had used in Syn-Aud-Con classes) is now a collector's item.

Don't say we didn't tell you. Even more relevant, to our thinking, is the joy and excitement of having "been in at the beginning" of a technology that will affect our cultural heritage through more faithful recording of great works of art.

dBV

In the IEEE dictionary we find *dBV* (see voltage level) and under voltage level..."The voltage reference normally used in standards is 1.0 volt"(not the value of 0.775 volts so often suggested in the popular audio periodicals). Again, it is not 1.0 volt RMS but 1.0 volt peak-to-peak that is most frequently employed. If a sine wave test signal is being used, then 1.0 volt p.p. is:

$$.707\left(\frac{1.0\text{Vpp}}{2}\right) = 0.354 \text{ V RMS}$$

Peak-to-peak is not mandatory, therefore, Syn-Aud-Con recommends the use of the dBV under the following set of conditions:

1. Make the reference 1.0 volt RMS
2. Use it only to measure "open circuit" voltages
3. Always carefully state the first two conditions so as to avoid the confusion generated by lack of standardization

The dBV is a most useful way to rate the output level of dynamic microphones, phono cartridges, and other transducers. A 150Ω microphone rated, for example, at -60 dBV re: 1 pa. is easily reconverted to voltage by:

$$E_o = 10^{\left(\frac{-60}{20}\right)} = 1.0 \text{ mv.}$$

and using:

$$G_m = 20 \log E_o - L_{pt} - 10 \log_{RMS} + 24 \text{ dB} = (-60 \text{ dB}) + (-94 \text{ dB}) - 10 \log 150 + 24 \text{ dB} = -151.76 \text{ dBm}$$

To which you merely need to add *your* performer's level (i.e., 115 dB) and you then have your available input power at the mixer imput:

$$-151.76 \text{ dBm} + 115 \text{ dB} = -36.76 \text{ dBm}$$

FURTHER NOTES ON THE DESIGN OF CENTRAL CLUSTERS

A central cluster is an array of sound sources clustered as close together as possible in the attempt to approximate the performance of an ideal single source device in the same location when such an ideal device is non-existent.

Since the sound pressure level (L_p) at the listener's ears from a single source equals:

$$L_p = L_w + 10 \log\left(\frac{Q}{4\pi(D_x)^2} + \frac{4}{S\bar{a}_m}\right)$$

where: L_w is the total acoustic power level

$$L_w = 10 \log\left(\frac{X \text{ watts}}{10^{-12} \text{watts}}\right)$$

D_x is the distance in meters from the sound source to the listener

Q is the directivity factor of the sound source (dimensionless)

$S\bar{a}_m$ is the total absorption in square meters (metric sabins)

$$S\bar{a}_m = s_1 a_1 + s_2 a_2 + \ldots . s_n a_n$$

It can quickly be seen that the ratio of the direct sound field level is dependent upon L_w, Q, and D_x and that the reverberant sound field level relies on L_w and $S\bar{a}_m$.

Further, L_w will be a function of the number of devices (N) in the cluster (NL_w) and their relative electrical input powers (W_e) but not the Q of such devices nor their distance from the listener.

$$L_{wt} = 10 \log\left(10^{\left(\frac{L_{w1} W_{e1}}{10}\right)} + 10^{\left(\frac{L_{w2} W_{e2}}{10}\right)} + \ldots . 10^{\left(\frac{L_{wn} W_{en}}{10}\right)}\right)$$

Finding L_{wt} accomplishes the same result as find 'N' for a cluster and using the Hopkins-Stryker equation to find the change in level between a point in the direct sound field and another point in the reverberant sound field (i.e., the ratio of direct-to-reverberant sound levels).

$$\Delta D_x = 10 \log\left(\frac{Q}{4\pi(D_x)^2} + \frac{4N}{S\bar{a}_m}\right)$$

where 'N' is calculated from equation six in Syn-Aud-Con Tech Topic Volume 5, No. 5.

$$L_w = L_{sensi} + 20 \log\left(\frac{D_{ref}}{.283\text{m}}\right) - 10 \log Q$$

For sound systems using a sensitivity rating (L_{sensi}) of xdB at a D_{ref} = 4' from 1 watt electrical input power the equation becomes:

$$L_w = L_{sensi} + 12.7 \text{ dB} - 10 \log Q$$

Knowledge of these parameters allows easy calculation of the ratio of direct-to-reverberant sound levels for inclusion in the more advanced Peutz equations that utilize

L_D, L_R L_{amb}, and RT_{60}

$$L_D = L_{sensi} + 20 \log\left(\frac{D_{ref}}{D_x}\right) + 10 \log(W_e)$$

L_{amb} is either calculated or measured

$$L_R = L_{w+} + 10 \log\left(\frac{Q}{4\pi(D_x)^2} + \frac{4}{Sa}\right)$$

RT_{60} is either calculated or measured

These further notes were inspired by material Dr. Eugene Patronis of Georgia Tech was kind enough to share with us relative to an AES paper he gave in May, 1981, at the Los Angeles convention.

SMILE

Brilliant people talk about ideas;
Average people talk about things;
Small people talk about people.

CONSTANT VOLTAGE OR CONSTANT CURRENT

There is not a necessity for a sound *system* engineer to concern himself with internal circuit design (though it is an enjoyable side skill in dealing with component malfunctions) but he should be conversant with the interface circuitry *between* components.

COMMON INTERFACE CIRCUITRY

Today, the most common type of interface is that of the preceding component at a very low impedance connected into the following component at a very high impedance (relative to the first unit). This concept has gained popularity, thanks to its increased signal-to-noise ratio and its simplicity.

One handicap is that it does not easily allow the use of "passive" devices between components and *forces* a buildout and termination if the passive device is to operate properly. Such circuits are referred to as "constant voltage."

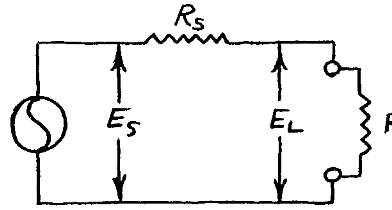
In measuring work, because of the omnipresence of voltmeters, it is often quite useful to operate a "constant current" system wherein voltage fluctuation directly follows impedance fluctuations.

The Shure Vocal Master system employed a "constant current" output system with great success.

"Impedance matching" *does not* mean that the sending impedance has to be the same as the receiving impedance in a link circuit. Impedance matching means that you are *properly* matched. A proper impedance match can be 130Ω to 15,000Ω or .001Ω to 8Ω or quite a variety of other values.

CONSTANT VOLTAGE OR CONSTANT CURRENT?

THE OUTPUT CIRCUITRY OF ANY SYSTEM CAN BE CHARACTERIZED BY THE DIAGRAM BELOW.



$$E_L = E_S\left(\frac{R_L}{R_S+R_L}\right)$$

CONSTANT VOLTAGE

LET $E_S = 1.0V$, $R_S = 0.01\,\Omega$

CALCU. $R_L = 8\Omega$, THEN $R_L = 16\Omega$

$$E_{L_1} = 1.0\left(\frac{8\Omega}{0.01\Omega + 8\Omega}\right)$$

$$E_{L_2} = 1.0\left(\frac{16\Omega}{0.01\Omega + 16\Omega}\right)$$

$$20\log\left(\frac{E_{L_2}}{E_{L_1}}\right) = 0\,dB$$

VOLTAGE DID NOT CHANGE WITH CHANGE IN LOAD VALUE

CONSTANT CURRENT

LET $E_S = 1.0V$, $R_S = 10{,}000\,\Omega$

CALCU. $R_L = 8\Omega$, THEN $R_L = 16\Omega$

$$E_{L_1} = 1.0\left(\frac{8\Omega}{10{,}000\Omega + 8\Omega}\right)$$

$$E_{L_2} = 1.0\left(\frac{16\Omega}{10{,}000\Omega + 16\Omega}\right)$$

$$20\log\left(\frac{E_{L_2}}{E_{L_1}}\right) = 6.02\,dB$$

VOLTAGE CHANGED IN DIRECT PROPORTION TO LOAD CHANGE

In the IEEE dictionary, "impedance matching" says see "load matching." Under "load matching" we find "circuits and systems: the technique of either adjusting the load-circuit impedance or inserting a network between two parts of a system to produce the desired power transfer *or* signal transmission."

Therefore, in most systems the term impedance matching really means connecting impedances together properly not necessarily of equal value. The illustration allows an insight into two common cases you easily might encounter.

One final thought. We agree with the circuit designers that the signal transmission path need only handle voltage or current with minimal power *until* you go back to the acoustical signal. You then *must* develop the acoustic *power* required.

Always remember, the function of a sound system is to distribute *equal acoustic power* to each listener while reducing to a minimum the amount of power sent to non-listener areas.

VISIT THE SYN-AUD-CON SEMINAR CENTER?

We are receiving a steadily increasing number of requests from 4,000 graduates to visit the Seminar Center when they are in the vicinity.

We greatly regret that such visits are not possible. The problem lies in our isolated location. In order to bring a visitor in (remember we are in a wilderness area behind locked gates) requires that a member of the Syn-Aud-Con staff and a vehicle spend 1/2 hour travel time to the locked gate, another 1/2 hour back to the Center, another 1/2 hour back down the mountain to return the visitor to the gate, and a final 1/2 hour returning to the center.

Naturally, the visitor wishes to see all of the spectacular hidden valley, its facilities, and parts of the surrounding forest. The result is that an entire day can be devoted to what, on the face of it, is a simple request.

We sincerely regret that we can't extend this courtesy to you should you come west on other business. We appreciate your understanding, and we hope you will have the opportunity to enjoy a Syn-Aud-Con class at the Center. When you do, you'll fully understand our problem with trying to bring you up one at a time.

DECADE CALIBRATION

In our last Newsletter we discussed "octaves" and the mathematics that allow easy calculation of intervals to the base two.

Now, let's talk about decades. A decade in history is ten years. A decade in audio is any 10 part interval (decade resistance boxes have controls calibrated from 0 to 9 on each knob). A decade in frequency would be defined as:

$$\frac{H.F.}{L.F.} = 10^1 = 1 \text{ decade}$$

where: H.F. is the highest frequency
L.F. is the lowest frequency

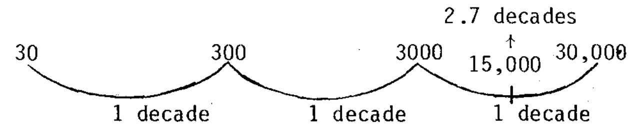
This means that we can write a general case expression for the calculation of how many decades there are in a given bandpass by:

$$\frac{H.F.}{L.F.} = 10^{(X \text{ decades})} \qquad \text{and:} \qquad \frac{\ln H.F. - \ln L.F.}{\ln 10} = X \text{ decades}$$

EXAMPLE

Using a frequency span of 30 to 15,000 Hz as we did in our last Newsletter in discussing octaves, we can now see how many decades that represents.

$$X \text{ decades} = \frac{\ln(15{,}000) - \ln(30)}{\ln 10} = 2.7 \text{ decades}$$



The question might arise in a different fashion, such as "What upper frequency limit would I have if I extended from 30 Hz 2.5 decades?".

$$H.F. = e^{((X \text{ decades } (\ln 10)) + \ln L.F.)} = 9{,}486.8 \text{ Hz}$$

or: What if I wish a 2.5 decade span with an upper limit of 5,000 Hz? What is my low frequency cutoff?

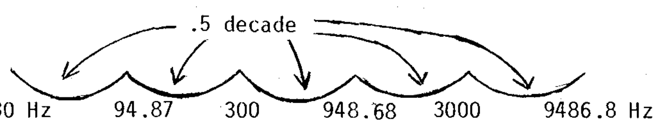
$$L.F. = e^{(\ln H.F. - (X \text{ decades})(\ln 10))} = 15.8 \text{ Hz}$$

Remembering that division into decades is electronically convenient (witness 1/10 decade rather than 1/3 octave filter designs), the general case equation for equally spaced logarithmic intervals can be derived from our approach above.

$$\left(\frac{H.F.}{L.F.}\right)^{1/N} = (\text{multiplier})$$

where: N is the number of intervals of equal logarithmic spacing (i.e., on a log scale are equally spaced)

Suppose, for example, that, using our earlier data, we want each frequency from 30 Hz to 9,486.8 Hz spaced 1/2 decade apart. That means we want a N = 5.

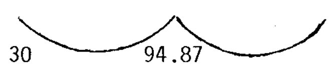


Using our equation, we find: $\left(\frac{9486.8 \text{ Hz}}{30 \text{ Hz}}\right)^{\frac{1}{5}} = 3.16$

Then our first mark is 30 Hz:

30 X 3.16 = our second frequency

and 3.16 X 94.87 = our third frequency



300 X 3.16 = our fourth frequency

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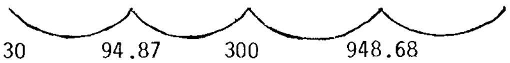
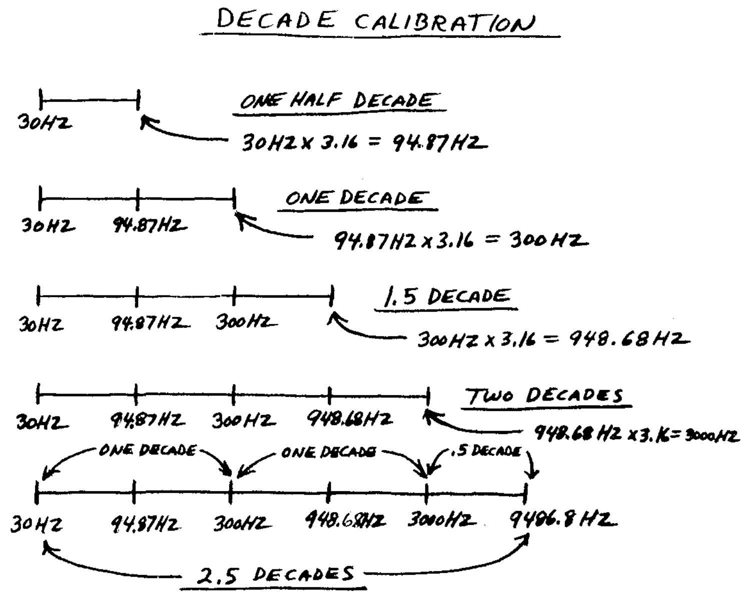
DECADE CALIBRATION continued



You can easily guess our next frequency.

and our last frequency as we complete 2.5 decades spaced at 1/2 decade calibration.

Finally, in amplitude levels, a ten to one increase or decrease is referred to as "an order of magnitude" and follows the same rules.



METRIC REPORT

David T. Goldman of the National Bureau of Standards has written a special report, "The Metric System: Its Status and Future," in the April 1981 IEEE "Spectrum." The first paragraph is interesting:

> "Progress toward metric conversion in the U.S., although less advanced than some would prefer, is occurring even without an enunciated national plan."

"A report by the National Bureau of Standards in 1971, 'A Metric America: A Decision Whose Time Has Come,' proposed the voluntary phase-in of the metric system of measurement over a 10 year period. That period is now almost over, but it is clear that the metric system *is not* about to become the predominant measurement system in this country."

The brainwashing of students from grade school on in the supposed benefits of the metric system is discussed without the recognition that most of these same students, as they mature, see through the propaganda and reject the training in favor of the ever present practical American system.

Nowhere in the article is the *fact* mentioned of the *inferior* metric standards we would have to replace our American standards with or the completely dishonest metric practice of showing the student *exact* conversion values while giving the scientist the correct conversion values.

One feature of this article that I did like is the complete listing of each derived quantity in terms of S.I. base units.

power watt *j*/s $(M^2 \cdot KG \cdot S^{-3})$* $\quad$ * $\frac{M^2 \cdot KG}{S^3}$

Freq Hz (S^{-1})* $\quad$ * $\frac{1}{S}$

Where: M is meters
KG is kilograms
S is seconds

The heartening news is that we are no longer associated with any backward nations in our opposition to metric measurement. We are now the *only* country not committed to it.

Syn-Aud-Con does not feel it is accidental that the nation with the greatest freedom of choice is the only one to still reject this obsolete French fetish. There are no perfect societies, at present, on this rotating ball of mud, but ours sure beats whatever is in second place.

The rights of free men risk abuses. Conformity is in direct ratio to loss of freedom. Freedom of religion can lead to a Jonestown or the constitution of the United States. The right of free men to bear arms can lead to a Hinckley or a Sergeant York. The right to be free of academic political control can lead to chaos or a new industrial revolution based on superior standards.

The very basis of freedom is the right to more than one way to solve problems. If we must someday spend trillions for a new system, let's first generate a better one than this hodgepodge of compromises between French mismeasurement of the earth's quadrant and their failure to force Mother Nature to operate solar time on a decimal base.

REVERBERATION

We have remarked in the Newsletter a number of times regarding the importance of W. B. Joyce's work in re-examining the fundamental reverberation equations. Now a colleague of Joyce, E. N. Gilbert, also of the Bell Telephone Laboratories, Murray Hill, New Jersey, has published what should be the final authoritative word on the subject.

> "Iterative Calculation of Auditorium Reverberation" in the JASA 69 (1) January 1981, pages 178 - 184. Quoting from the abstract:
>
> "Formulas for reverberation time have always been derived assuming a constant distribution function for sound intensity throughout the auditorium. The true distribution is known to satisfy an integral equation, first derived by Kuttruff, which has not found much application because it contains the unknown reverberation time itself as a parameter. Here energy considerations supply another relationship which, together with the integral equation, form the basis of an iteration procedure. The intensity distribution and the reverberation time are both corrected on each iteration....."

The rapid convergencies possible with Gilbert's iterative technique offers a powerful new tool for the prior analysis of an auditorium in advance of its construction.

SOME RANDOM THOUGHTS ON REVERBERATION EQUATIONS AND THEIR APPLICATION

Syn-Aud-Con has, since the acquisition of TEF™ measurements, become increasingly critical of when and where to use equations derived from the basic reverberation assumption. The outstanding misuse, obviously, being the measurement and calculation of reverberation times in recording studio control rooms that *do not have a reverberant sound field*. In today's control rooms the energy that might have produced a reverberant sound field is far below the ambient noise levels present in the space and, therefore, what one measures is merely the transient time of some early reflection.

Again, Syn-Aud-Con's experience in real world measurements using Energy Time Curve (ETC) apparatus is the realization that the concept of reverberation time is but a coarse approximation of the quantity and quality of the ratio of direct-to-reflected energy present in a given location.

ACCEPTABLE LISTENING CONDITIONS

When the following conditions are met, a given location is judged as an acceptable listening environment:

1. A good signal-to-noise ratio. (We are increasingly inclined to believe that this means the direct sound level to noise ratio.)

2. A correct ratio of direct-to-reflected sound energy, particularly in temporal terms. (Beranek's early identification of the initial time delay gap is now coming into much fuller recognition.)

3. Sufficient acoustic gain.

Sabine's genius is clearly apparent in the very first pages of his published papers when he writes:

> "In order that hearing may be good in any auditorium, it is necessary that the sound should be sufficiently loud;
>
> "that the simultaneous components of a complex sound should maintain their proper relative intensities;
>
> "and that the successive sounds in rapidly moving articulation, either of speech or music, should be clear and distinct, free from each other and from extraneous noises."

Let's dissect Sabine's criteria for good listening.

> "In order that hearing may be good in any auditorium, it is necessary that the sound should be sufficiently loud;" -- A good signal-to-noise ratio which implies adequate acoustic gain.
>
> "that the simultaneous components of a complex sound should maintain their proper relative intensities;" -- Proper equalization either of the sound system or of the reflected sound so that no extraneous source of energy cancels the direct sound from the source.
>
> "and that the successive sounds in rapidly moving articulation, either of speech or music, should be clear and distinct, *free from each other* and from extraneous noises." -- Sabine clearly sensed the *temporal* basis of interference with direct sound.

Continued on next page

CALCULATING RATIOS

Thus, calculating is of paramount importance in determining

1. The direct-to-reverberant ratios, L_D/L_R
2. The signal-to-noise ratios, L_D/L_{AMB}
3. The inclusion of the temporal spacing of significant energy levels during the first 50 msec
4. The suppression of significant individual energy levels after the first 50 msec

S/N ratios and L_D/L_R ratios are adequately defined in Peutz's important articulation equations. We also know a great deal about temporal spacing thanks to Henry, Haas, and Kuttruff's work.

1. No energy level after L_D should be greater than L_D
2. $L_R < L_D$ and spaced optimally 20 msec beyond L_D with each subsequent L_{R1}, L_{R2}, etc., spaced 5 to 10 msec further out in time
3. Beyond 50 msecs no energy levels should exceed the desired L_D/L_R ratio
4. The greater the density and the closer to exponential level change with increasing time the mass of late L_R's can be made, the more diffuse the reverberant sound field.

SUMMARY

Today, with ETC measurements the engineer fortunate enough to be gaining experience with such apparatus is rapidly turning from classic reverberation time equation calculations to specific correction of temporal misalignments, adjustment of spurious energy returns via specific interruptions of their specific paths, and is enjoying the thrill of understanding and significantly controlling the complex interactions between acoustic energy and the boundaries containing it.

We are struck by the intuitive sense of Fitzroy (who realized the influence of the three principal axes on the decay rate of sound), Leo Beranek with initial time delay (ITD), and Sabine, who saw it all in his basic work but was forced to lump it into a practical all-in-one measurement.

ERRATA

Newsletter Volume 8, No. 2, had a few typos (all but one made after Helen Range proofed) but the most glaring (and funny) mistake was "Errata" spelled "Eratta." What does one say?

Page 8: *Digital Time Delay* — "Sound Path Difference" from Industrial Research Products - the headings "Delay msec" and "Feet" are reversed. Instead of 4 msec of delay = 3.5 ft; 3.5 msec = 4 ft. The formula is 1 ft = .885 msec of delay, which allows you to figure the delay for any distance.

Page 11: *An Overlooked PZM™ Advantage*

"Diaphragm Size"
.57", .11", .057" should have read .57' or 6.8"; .11" should have read .11' or 1.32"; .057" should have read .057' or .6"

This is a typo that pleased us in that it assures us the Newsletter is being read judging from the number of people who called the mistake to our attention. We didn't think the Newsletter had time to be received when we heard from JIM FULLMER, a Consultant in Salt Lake City, and DON MEREEN of Telex. And they just kept coming.

Page 11: *%AL_{cons} HP41 Documentation*

RANDY GAWTRY called to say $AL = 100\left\{10^{-5(A+BC)-ABC)}+0.15\right\}$

should have read: $AL = 100\left\{10^{-2((A+BC)-(ABC))}+0.15\right\}$

JOHN LANPHERE sent in the same correction.

MARCH 1981 CLASS

We're sorry that our local photo shop lost a roll of film covering our March seminar.

"ARTICULATION" TEST

Theophilus Thistle, the successful thistle sifter,
while sifting a seiveful of unsifted thistles
thrust three thousand thistles through
the thick of his thumb.

Original Author Unknown

Thank goodness it's not mandatory to be able to say this one rapidly in order to work in audio. It is an excellent test that quickly reveals a "too hot" high frequency response. (Given to us by ROBERT BOYD of the Edmonton class 1976.)

THE DIGITIZATION OF AUDIO BY PETER SUTHEIM

Peter Sutheim so succinctly expresses Syn-Aud-Con's feelings about the present state of the digital recording art that we asked for and received permission to reprint his comments as they appeared in a journal called *Audio Amateur*.

THE GROUNDED EAR

by Peter Sutheim

THE DIGITIZATION OF AUDIO

BY NOW A TOUCH of weary cynicism afflicts every observer of the audio marketplace Many remember personally the cycles of fascination and disenchantment with revolutionary breakthroughs like solid state and quadriphonics Each promising development, exploited to the utmost in the marketplace, seems to have left a kind of hangover in its wake So if your attitude toward the digital *tsuris* is ho-hum—another revolution,' let me suggest that this one is enough different to claim your serious attention Granting acknowledged theoretical and practical advantages of digitized audio, growing evidence signals serious trouble, and bad news warrants closer attention before nearly irrevocable steps toward standardization are taken

I want to devote this column to a fair presentation of the cases for and against digital sound at the present state of the art No serious journalism exists without some bias The slant of this examination is that while the advantages of digital audio have been published enthusiastically and repeatedly in uncountable variations in the technical journals and the popular media, its drawbacks, at least in its present forms are not nearly as widely known This column is offered as a step toward correcting that imbalance and perhaps a stimulus to a wider and more fundamental discussion of digitized audio

I haven t space here for much detail about how digital encoding, storage and recovery work for audio at the end of the article are references for further reading A brief refresher may be in order, however The governing principles are *sampling* and *quantization* The program signal is sampled for its intensity at regular intervals, typical clock rates now being in the vicinity of 50,000 samples per second Each resulting voltage sample is then binned by an analog-to-digital (A/D) converter according to its intensity in one of a large number of possible levels, each represented by a binary digital "word, made up exclusively of ones and zeroes A sample of given intensity belongs in one box or another, not in none and not in two Thus is the audio program encoded in digital form, wherein it can be recorded on tape or disc, stored in electronic memory or transmitted by the same means as other digitized data

For playback, a digital-to-analog (D/A) converter evaluates each binary word at the clock rate and accordingly produces a voltage pulse of the appropriate level These pulses are integrated to recreate the original audio waveform

The conception is a brilliant example of abstract thinking, and mathematically unimpeachable The original program can be dissected and reassembled with any desired accuracy depending only on the twin choices of sampling rate and number of quantization levels It is there, and in the physical realization of digital systems, that difficulties begin

The technique is attractive in the practical world of recording and signal-processing for several reasons

1. Information stored in the form of binary bits is less vulnerable to contamination by broad-spectrum noise (e g white noise) or to put it in a folksy way it ought to be easier to build a circuit that can tell whether or not a pulse is present in a given spot than one that has to deal with all the subtleties of a musical signal This attribute applies to long-term storage, accidental partial erasure, print-through, crosstalk, etc
2. By similar reasoning, when copying digital tape in digital form (without decoding), signal-enhancement techniques can produce a copy that is for all practical purposes identical to the original The prospect of unlimited copies each as good as the master is certainly appealing
3. The frequency response and dynamic range of a digital recording system are limited only by the choice of sampling rate and by the word length (number of bits per word), which together establish the resolution, or information-carrying capacity, of the system Frequency response down to zero Hz (direct current) is an accomplished reality, as is a dynamic range of 90dB
4. If the clock rate in the recorder is stable, and if the playback clock is controlled by pulses put on the tape at the time of recording by the same clock that controlled the initial sampling, then flutter and wow—the consequences of speed variations in the tape or disc transport—disappear
5. Error-correcting codes can be devised which, by adding an extra bit or two of information to each word, can compensate for tape dropouts (loss of contact with the head or imperfections in the tape itself)
6. Digital recordings can be edited very precisely by purely electronic means, without resorting to the comparatively crude and fussy technique of physically cutting and splicing the tape "Trial edits can be made evaluated and remade in much less time than cut-and-splice allows
7. Digitization makes possible some highly sophisticated signal processing, such as artificial reverberation (removal as well as addition!), precise filtering of unwanted noise, correction for colorations, etc Once the program has been digitized, it can be operated upon by a suitably programmed computer in the same way as any other digital data

The first three items of the list are the ones with the most general appeal continually promoted in the popular media None of the claims made for digital sound—at least as they have been summarized here—are untrue They (and their promotion) reflect rather, an idealized conception which is betrayed to some degree by the available hardware In what follows, those shortcomings will be examined, and in the last parts of this column we ll have a look at some perceptual discontent with digitized audio that is not so far accounted for by the theory or the specifications

Opposition to digital sound seems to fall into two camps those who hear something wrong with it and those who don t In the first category are people who seem to be responding emotionally to connotations—who recoil at the idea of "cutting beautiful music up into little bits," as well as people who feel, often vehemently and apparently quite sincerely, that the digitized music they have heard simply does not sound as beautiful as it ought to Some of that latter subgroup have a technical understanding of the physics and math of digital audio others grasp it in an intuitive way Some have expressed feeling actual physical discomfort during digital auditions

The second category comprises people who have heard satisfactory (or at least promising) reproduction from digital sources but are opposed to standardizing it in its present form feeling, for various reasons that it may harbor latent flaws that will haunt us later They sometimes cite color television and FM stereo as examples

Straddling both categories are people who have become aware as well that digital recording has sometimes failed to deliver on its most highly touted features The unlimited perfect copyability, for example, does not always work in certain systems, because the error-correction scheme cannot cope with the accumulated dropouts in successive transfers The resulting uncertainties in quantization create audible "holes or extremely annoying forms of noise or distortion peculiar to digital systems

Three incompatible digital recording formats are on the market, differing in almost every possible way tape speed, word length, sampling rate, error-correction scheme and other details This fact is in part responsible for the current scramble toward standardization, naturally, every proponent would like to see its format become the standard and is therefore quick to point out the flaws in the competition while defending its own

Defects or limitations of one design do not necessarily apply to another Only a devoted digital-watcher will be able to keep it straight, for most of us, the claims and attacks are homogenized into an ominous fog, in which it seems likely that an eventual standard will be concocted as much through industry politics and economic clout as by a rational analysis of merit

Such a climate is hospitable to rumors, some of which I have heard from two or more seemingly independent sources but have been unable to confirm to my satisfaction An example is the story of the digital tapes in record company X's vault which, though only two or three years old, are forever unplayable because the digital codes on the tape are too badly "smeared" magnetically, supposedly due to some combination of print-through and self-erasure Whether this, if true at all, was due to an accident or to some essential flaw in the system, we may never know

Continued on next page....

Such an event can be tragic, a lot or a little, depending on the amount of wasted effort and lost art, but there is a deeper, in fact quite fundamental, issue with more far-reaching consequences.

Several authorities[1,2,3] agree that the present electromechanical recording system, with the vinyl phono disk as its product, is unsurpassed in its information-storing capacity and its longevity, when correct methods for playback and preservation are used. To paraphrase this convergence of opinion: we know how to make records that are better than we can play at the moment. The limitations are in playback styli, cartridges, tonearms—not in the disk-cutting or pressing process. (The *Popular Electronics* article[3] is worth reading. It arrives at a dynamic range of 88dB for a top-notch release pressing, computing the noise floor against the tracking limit of a high-quality cartridge, and compares this to the 84 to 85dB claimed for one digital disc system being proposed, and to the 90dB claimed for digital tape systems. The article also states a bandwidth up to 45kHz for conventional vinyl discs.)

Digital encoding is a very different matter. *The choice of encoding format establishes forever the maximum information content of the recording medium.* When one chooses a sampling rate and a quantization format, one is in effect saying, "I judge this to be a sufficiently accurate replica of my original."

The choice of sampling rate *may* be based on faulty, possibly outmoded assumptions about hearing. It is a textbook commonplace that the upper limit of human hearing is approximately 15 or 16 or 20kHz, depending on the age, sex and other individual factors of the subject under test. The figure is used to justify a sampling rate of, in one case, as low as 32kHz. (Information theory requires a sampling rate at least twice the highest frequency to be encoded.)

Yet, Fourier analysis notwithstanding, the music of the world is not described entirely by clusters of sine waves. Normal hearing can respond to pulses as short as 5 *micro*seconds' duration—five times faster than one half-sinusoid "pulse" in a 20kHz sine wave. Since the acoustic universe has not agreed to end at the 20kHz marker established by steady-state sine wave measurements, it is quite likely that information of a sort is there which contributes to the lifelikeness of the best recordings.

Admittedly, it is probably a rare microphone that can respond that high (though it has been suggested that the unique transparency of certain ribbon microphones is attributable to their gentle 6dB/octave rolloff at the highest frequencies, in contrast to other microphones with much steeper high-frequency boundaries). But microphones evolve, too, and we have already seen the fallacy of letting the limitations of one part of the chain determine deliberate limits in another.

Without being explicit enough, the Discwasher people[3] write, "... there is some evidence that trained listeners sense the absence of such [extremely high] frequencies if the audition period is sufficiently long. Studies showing the listeners do not miss anything above 15kHz have used relatively short auditions." One such test is presented in detail in reference 4, using paired quasi-musical pulse recorded music samples of 1-second duration. With seven different types of sharp-cutoff low-pass filters to limit the bandwidth presented, the 43 subjects, considered as a group, did not reach the threshold of discrimination for this type of test (75%, or halfway between chance—50%—and perfect score—100%). Whether certain individuals were able to discriminate more consistently is not reported.

One curious fact emerges from a table of results: the 21 professional sound engineers among the subjects achieved quite noticeably *lower* scores (poorer discrimination) than the 22 non-engineers—on one filter, lower by ten percentage points. The paper does not comment on that aspect of the result.

The choice of quantization levels, which determines the number of bits per word, may likewise be based on an incomplete understanding of our hearing. A 16-bit system, the largest number so far applied commercially, gives a theoretical 95dB dynamic range with more than 65,000 level steps, which seems almost astronomically vast. Yet most of that capacity remains unused most of the time. Material near the threshold of audibility—the last reverberant sound decaying into silence, or the weaker harmonics of a complex musical note—may be encoded with only 2 or 3 bits.

The *quantization noise* resulting from a random hunting between quantization levels begins to be significant at these low levels, in much the same way as class-B crossover distortion in amplifiers is more audible at low levels than high. Unlike the more-or-less constant noise floor of conventional recording techniques, quantization noise exists only when signal is present (rather like modulation noise in analog tape recording). It is thus possible to claim for digital recording an essentially silent background—and to hear no hiss—and yet to experience a signal-dependent noise (or distortion).

Could this account for some listeners' reports of an "unnatural" quality to the decay of sounds recorded digitally? This problem is compounded by re-recording a digitized signal digitally after it has been already restored to analog form. Various ingenious methods have been applied to redistribute the channel capacity more advantageously: floating-point encoding and some forms of compression and expansion rather like familiar methods used in analog work.

Alan Sides, a recording engineer known for his work with pop artists Diahann Carroll, Kenny Burrell, Billy Preston and others, has done several albums with digital technology using each of the three commercial systems (3-M, Sony and Soundstream) and finds some of the results baffling and unsatisfying. "It was like the equalizer didn't work properly. ..." he said in an interview. "If I boost 16k it sounds like I'm boosting 10k... When you first play it back it seems fairly close to the original, but when you play with it a little bit longer and you start mixing with it, the high-frequency information sounds a little peculiar." His comments were based on work with the 3-M machine.

He also reports that while the Sony PCM-1600 "sounds awfully good" on originals, after a couple of digital-to-digital transfers "there seems to be just slightly less echo each time down." Stranger still is his report of comparing the Sony digital original with a simultaneous 30-ips analog master on an Ampex ATR. The digital master sounded much better—then. But a direct 30-ips analog copy of the digital original "wasn't even close" to the Ampex analog original. However, *digital copies* sound better than the 30-ips original!

"This is something I just don't understand." He conjectures, though, that the excellent midrange clarity in a digital original keeps one from noticing the absence, or the "peculiarity," of the extreme highs. When the digital tape is transferred to a conventional analog machine, which smears things a bit, then the high-frequency strangeness is evident.

He has been dissatisfied also with edited digital masters from the Soundstream system. (Such a product would presumably be two digital generations off the original—once into the editing computer, then back onto tape.) "Less ambience, duller" were the words he used in comparing it to the original.

Recording engineer Rick Ruggieri (Neil Diamond and others) has been through four albums with digital. "None of them have really come out like they started," he reported. He heard sonic changes serious enough to change the balance of the music, although he doesn't really object to the sound. His main objection is his feeling of too little control over the steps in the process, of not knowing where something is going awry. (He worked with Soundstream. He has heard a Sony PCM-1600 and slightly prefers it, but has not worked with the Sony editing process.) "It's still a baby," he says. "I'm not knocking it, but [right now] the outcome is not worth the expense.... They're trying to shove it down everybody's throat."

Curiously, every engineer I talked to had had his share of digital distress, but spoke of colleagues who told "hours of nightmare stories" of their experiences with digital sound. Some of the stories, like the one of the deteriorating archives mentioned earlier, are difficult to track down. In a typical pattern, someone who is reported to have had unfavorable experience is questioned, relates one or two anecdotes and then refers to someone else who's *really* had a bad time! (lost tracks, whole songs lost, hopeless degradation of quality).

This is more sociology than audio, but what is clear is that the conquering wave of digital audio as viewed from the trade and popular press is being countered to some degree by a lusty undertow of disgruntled and wary recording engineers. Are they a miniority? Probably. Is that why so little of this discontent has surfaced in print?

I mentioned in passing that the sampling rate must be at least twice the highest frequency component present in the signal. This is a basic principle of information theory; a proof can be found in many textbooks, but is beyond the scope of this survey. The principle has a serious consequence that is not immediately obvious: the need to suppress, before digitization, any energy at a frequency higher than one-half the sampling rate. Higher frequencies will be misinterpreted by the encoding process in such a way as to produce a spurious, inverted, mirror-image spectrum, and highly audible distortion.

This is called *aliasing*, and the need for suppression requires *anti-aliasing filters* to be included in the A/D conversion equipment. For economic and bandwidth limiting reasons, it makes sense to keep the sampling frequency as near to twice the highest audio frequency as possible; that in turn demands filters with very sharp cutoffs. Such a filter, which, for example, might have its 3dB "knee" at 15kHz and be better than 60dB down at 22kHz, manifests some pretty dramatic phase shift along with its amplitude characteristic. Some suggest the anti-aliasing filters may be as much to blame as the digitization process itself for audible high-frequency aberrations.

Whether or not that's true could be determined by inserting such filters into a conventional high-quality analog chain and doing a controlled listening test. Something like this was done in the experiment cited in reference 4, although the effects of the filtering and the digitization could be synergistic. That experiment was conducted with filters only.

Some readers may be now be wondering why all the fuss—why not just choose some extravagantly high sampling rate and number of bits? The answer is only partly cost; the other part is that our modest, homely business of recording music accurately, so unspectacular when compared with satellite transmissions of video and high-speed data, is in fact bumping the limits of technology. A 16-bit A/D converter has to resolve level differences of about 150 microvolts over a 10 volt signal amplitude range, and, assuming a 50kHz sampling rate, would have to make its first (most-significant-bit) decision in about a microsecond.

The other surprising aspect of this is the relatively enormous bandwidth required. A 16-bit, 50kHz system, with an upper frequency limit of 25kHz, requires an 800kHz channel. Each additional bit doubles the bandwidth requirement, as does doubling the sample rate.

Consider, for comparison: Mark Levinson has claimed for his analog tape recording system, using a modified Studer A-80 at 30/ips, two tracks on half-inch tape and his own electronics, a noise-floor-to-saturation dynamic range of 90dB[5] In terms of recording fidelity only (not considering the possible further advantages of digital manipulation), this performance probably equals that of any current commercial digital system.

So it begins to be apparent that digitization of audio is not *necessarily* synonymous with an improvement in fidelity, but represents perhaps mainly an improvement in the convenience of certain operations, such as editing and signal processing. And this improvement in convenience is obtained at a rather severe cost in bandwidth—which in practical terms manifests itself in the amount (area) of tape consumed per unit of recording time.

I close this portion of the survey with a quote from an extraordinarily intelligent book, *Applications of Digital Signal Processing*[6] "What would be the bit rate of a converter that was perfectly matched to the human auditory system? We can consider this question with respect to all audio signals covering the complete dynamic range and hearing bandwidth or with respect to the smaller class of signals referred to as natural music. In both cases, the optimum converter will be one with the minimum perceptual error. . .

"An optimum converter is one with minimum bit rate such that further minimization of the error between the converted and unconverted audio signal is inaudible. Attainment of this goal requires a good understanding of the psychoacoustic effects of various conversion errors. Unfortunately, there is no complete model of auditory perception, but only a large literature documenting various phenomena."

Subjective responses are not hard to come by in audio, and when a new development appears there is never a lack of opinion about its merits. But, as far as I can determine, there has never been a stranger evaluation of a new audio technology than one presented to the 1980 Los Angeles Convention of the Audio Engineering Society. Dr. John Diamond, a psychiatrist from Australia now living and practicing in upstate New York, delivered to an overflow crowd his findings on the use of digitized music in therapy.

He claimed to have treated thousands of patients and to have evaluated thousands of (conventional) records for their effects in reducing stress and promoting healing. To his surprise and dismay, upon trying digitally produced disks he discovered that they consistently *increased* the level of stress in the hearer, as manifested by temporary weakness in the deltoid muscle (a large muscle that runs from the upper arm into the shoulder).

Diamond called for volunteers from the audience and demonstrated his effect, first testing each subject by applying downward pressure on his or her outstretched arm, then testing again with a bit of music from a conventional (analog), record, and last with digitized music. The astounded volunteers, who had tested strong at first and with the ordinary recording, were unable to resist Diamond's moderate downward force while listening to the digitized production.

Individuals in the audience, which was by now in an uproar, heckled Diamond with shouts denouncing his "unscientific" presentation. He replied that he was not about to "do science" in a 20-minute presentation before a crowd of 300, and tried to remind his hearers that the convention was a forum for presenting observations and results. He encouraged others to pursue his findings. The presentation ended, after an exceptionally long and tumultuous question period, with many members of the audience angry, baffled, curious or amused, in various blends. (A summary of Diamond's presentation can be found in ref. 7.)

Since then, those who have been unwilling to dismiss Diamond as a joker or a fraud have speculated on possible physiological mechanisms to explain his findings. Some have conducted informal experiments to repeat his results—with equivocal outcome. One formal experiment by Nelson Morgan of the Electronics Research Laboratory of the University of California at Berkeley was published in the *Journal* of the Audio Engineering Society[8] Its results showed no correlation between digitization and stress as indicated by "arm pushing" (Morgan's phrase), basal skin resistance and galvanic skin response.

The published letter reporting that experiment elicited a response from Dr. Diamond which the AES has declined to publish, perhaps preferring to consider the matter closed. (The entire affair has been something of an embarrassment to the AES, which likes to think of itself as very scientific.) Diamond's critical reply, in which he reports a widespread disinclination on the part of digital equipment makers to cooperate with him, was published in the Fall 1980 issue of the *Syn-Aud-Con Newsletter*[8]

Syn-Aud-Con (Synergetic Audio Concepts) is Don and Carolyn Davis's audio education enterprise—rather more open than most to new ideas, having early embraced such now widely accepted techniques as time-delay spectrometry, pressure-zone miking and live-end-dead-end control-room design. They have defended and supported Dr. Diamond in his attempt to communicate his concern over possibly unhealthful effects of digitized sound.

In a letter of his own to the AES[9] Don Davis writes: "Mr. Morgan fails to state that his 'Comments' [ref. 8] was in fact financed by . . Pioneer and that the individuals generating the test tape are not without bias toward the outcome. Using anti-aliasing filters on the analog recording is inadmissible.

"Therefore, the evidence clearly points out:

"1. It is not an impartial attempt to evaluate Dr. Diamond's demonstration.

"2. The tests conducted are not relevant to the claims Dr. Diamond made."

Davis then quotes passages[10] indicating that subconscious "decoding" processes in the brain involving "biological clocks" are crucial to our sensory discriminations—hearing, particularly localization, as well as others. Davis' purpose in quoting these writings appears to be to suggest that something in digitized audio—perhaps the sampling? perhaps the quantization?—may interact unfavorably with our mental processes.

Davis points out also that Diamond's test of the deltoid muscle is a test of the subject's reflex motor response ". . [I]t is the subject's reaction time change that allows the arm to be pushed down without effort. It is not—repeat—not a test of strength."

This is a controversy close to the edge of science, reminiscent of UFO studies or attempts to demonstrate ESP. Because it is so deeply involved with the belief systems of the disputants, it may never be resolved to anyone's lasting satisfaction. I would let it all go at that if it were not for the surprising number of persons (among them Doug Sax of The Mastering Lab and Sheffield Lab records) who have independently reported puzzling and persistent discomfort when listening to digitized audio. Typical comments (paraphrased): "It sounds ugly—not beautiful," and "I can't stay in the room with it," and "normally we can master for hours and feel fine, but after a morning with a digital tape, we're just wiped out."

And so Dr. Diamond's plea in his response to Morgan's experiment is perhaps best heard as an alarm: "What if my test results are right? . Billions of dollars are involved in the digital process. Before we inflict unnecessary stress on the present generation of listeners and future generations, let us at least wonder that there may be a stress factor which may have been demonstrated at my presentation. Let us look, and let us work to overcome this. In so doing, we will in part repay our debt to music for all that it has done for us."

Meanwhile, in the technical journals, "standardization" is the cry of digital progress. □

REFERENCES

1. Doug Sax, talk before informal gathering of Los Angeles section of Audio Engineering Society, recorded 21 September 1980.

2. Mitchell A. Cotter, conversation recorded in Chicago for broadcast on "In Fidelity," KPFK, Los Angeles, 18 June 1980.

3. Bruce R. Maier, *et al.*, "Phonograph Playback—It's Better Than You Think," *Popular Electronics*, Nov 1980.

4. G. Plenge, *et al.*, "Which Bandwidth is Necessary for Optimal Sound Transmission?" *Journal* of the Audio Engineering Society, Vol. 28, No. 3, pp. 114-119 (March 1980).

5. Confirmed in unpublished letter from Louis Fielder, member of the research staff of the Recording Physics Section of the Advanced Technology Dvision, Ampex Corp., to James Boyk, who had inquired whether the claim was plausible.

6. Alan V. Oppenheim, editor, *Applications of Digital Signal Processing*, Prentice-Hall, Inc., chapter 2 (B. Blesser and J. M. Kates, "Digital Processing in Audio Signals," p. 48).

7. *Syn-Aud-Con Newsletter*, Vol. 7, No. 4, Summer 1980, p. 3 (Syn-Aud-Con, PO Box 1115, San Juan Capistrano CA 92693).

8. Nelson Morgan, "Comments on 'Human Stress Provoked by Digitalized Recordings,'" *Journal AES*, Vol. 28, No. 9, p. 613; reprinted in *Syn-Aud-Con Newsletter*, Vol. 8, No. 1, Fall 1980, pp. 29-30.

9. *Syn-Aud-Con Newsletter* (ref. 8), pp. 35-37.

10. G. H. Whitrow, *The Natural Philosophy of Time*, 2nd edition, Clarendon Press, Oxford, England, 1980 (reference quoted from *Syn-Aud-Con Newsletter*, ref. 8).

AES RESPONSE

The intellectual level of the AES is represented by Dr. Robert O. Fehr's letter of February 25, 1981. We publish it without comment as it speaks for itself.

AUDIO engineering society, Inc.
INTERNATIONAL HEADQUARTERS
60 EAST 42nd STREET, NEW YORK, N.Y. 10165 U.S.A.

Telephone (212) 661-8528 Telex: AES620298UW

PLEASE REPLY TO

Journal Office

1981 January 13

Mr. Donald B. Davis
Synergetic Audio Concepts
P. O. Box 1115
San Juan Capistrano, CA 92693

Dear Mr. Davis:

Thank you for your letter of December 15. The attached revised copy (now dated December 15) has been sent to Dr. Doi, Mr. Nelson, and Mr. Locanthi so that they will have the opportunity to send us a response to be considered for publication with your letter.

Sincerely,

Bob Fehr /pmm

Robert O. Fehr
Editor, Journal

PMM:dsa

AUDIO engineering society, Inc.
INTERNATIONAL HEADQUARTERS
60 EAST 42nd STREET, NEW YORK, N.Y. 10165 U.S.A.

Telephone (212) 661-8528 Telex: AES620298UW

PLEASE REPLY TO:

Journal Office

1981 February 25

Mr. Donald B. Davis
Synergetic Audio Concepts
P. O. Box 1115
San Juan Capistrano, CA 92693

Dear Don:

We have now received the replies from our reviewers on your letter of November 17 (revised December 15). On the basis of these reviews and further considerations and study, I have decided not to publish this letter. The reasons for not publishing are as follows (my comments are in the same sequence of your letter):

That there was a discussion on the possible problems caused by alternate sampling has already been mentioned in the Journal.

Mr. Morgan informed me that he followed Dr. Diamond's recommended procedure. There was no reason to consider other references.

In your letter you question the honesty of reported research without providing proof for this accusation.

Some of your remarks may be offensive to our Japanese readers.

Your references may be of general interest but our reviewers believe that they do not apply.

I trust that any committee working on digital standards will be aware of Dr. Diamond's test, and will critically evaluate them when writing a standard.

In addition, it is not a function of the AES to sponsor research. Regarding your suggestion for a panel, it would be most appropriate for you to approach those concerned directly.

Sincerely,

Bob /pmm

Robert O. Fehr
Editor, Journal

ROF:dsa

"SYN-AUD-CON"

SYNERGETIC AUDIO CONCEPTS / CONSULTING · SEMINARS, P.O. BOX 1115, SAN JUAN CAPISTRANO CA 92693 (714) 496-9599

March 27, 1981

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

Dear Mr. Fehr:

Before I comment on your letter of February 25, 1981, I would like to see the replies from your reviewers.

I believe this is customary. At least, you have done so previously.

Sincerely,

SYNERGETIC AUDIO CONCEPTS

Don Davis, President

DD/jk

It is doubtful if the AES has heard the end of Dr. Diamond.

Dr. Diamond has been invited by Herbert Von Karajan to address the Salzburg Music Festival April 21st. Following the Festival, Dr. Diamond has been invited to address musicians and music critics in Zurich, Munich and Frankfort.

Syn-Aud-Con's role is to encourage original thinking on the part of its graduates. We hope that the Dr. Diamond-AES affair has been instructive.

NEW TEF™ LICENSEES

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Hagenbach Labs
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Indianapolis, IN 46222

Mr. Steven Hodge
Steven Hodge Company
Box C8
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Mr. C. Michael Hoover
Audio Technical Services, Ltd.
239 Mill Street NE
Vienna, VA 21180

Mr. Glen Ballou
Sikorsky Aircraft
North Main Street
Stratford, CT 06602

Mr. Paul Jarvis
Compass Point Studios
P. O. Box N4599
Nassau, BAHAMAS

Mr. Don Wolford
Don Wolford Sound Systems
322 South Arroyo, Unit A
San Gabriel, CA 91776

ARTICLES OF INTEREST

"Examination of Audio-Bandwidth Requirements for Optimum Sound Signal Transmission" by Muraoka, Iwahara, and Yamada is the lead paper in "AES Journal," January/February, 1981.

> "As a result of the tests we found the critical cutoff frequency to be around 15 kHz......" and goes on to say, "Examinations were conducted at listening levels of 70, 80, and 90 dB sound pressure level."
>
> "Listeners' audible limits obtained from the 70-dB sound pressure level test are considered to be fairly accurate. *At 80 and 90 dB sound pressure level, however, some listeners conceded that judgments were made by perceptions of pressure, headache, or ear ringing, etc.*, and not by the test tone itself, leaving the results somewhat questionable." (Would the Japanese authors really ask us to believe that their test subjects get ringing in their ears at 80-90 dB?) Truly, their results are questionable.

Quoting from page 56 in the same journal, "Digital Audio Technical Committee Report," from P. Rodger's report:

> "It was her feeling that if signals up to 20 kHz were presented to listeners for test purposes, then the 20 kHz levels *would have to be higher than the threshold of 100 dB sound pressure level for their presence to be noticed.* (Italics mine) She further felt that the tests should be done in stereo to preserve the spatial characteristics of the natural sound and that by doing so the thresholds might be more critical." She earlier pointed out that bandwidth limit measurements involved monaural sources.

Again, quoting from "Examination of Audio-Bandwidth Requirements for Optimum Sound Signal Transmission":

> ".....bandwidth should be based upon human audibility. However, the determination should be guided by the audibility of the general public, not by the golden-ear extreme. This decision is significant not only for reasons of human engineering, but also for reasons of economy in resources and energy.
>
> "All things considered, a bandwidth of 15 kHz is considered to be most reasonable. Music producers, those who engage in synthesizers in particular, ought to be aware of this conclusion."

Quoting from Len Feldman's "Ambient Sound" in the April, 1981, issue of "Modern Recording":

> "Mr. Hans Fantel, who writes a column on sound for readers of the prestigious *New York Times*, prompted me to think about the subject of this month's 'Ambient Sound' column. In a recent column, Mr. Fantel, discussing the new 'digital' discs (which he correctly goes on to explain are not really 'digital' but only digitally mastered), goes on to say, 'Yet as such (digitally mastered) recordings proliferated in the latter part of the year, it became evident that nearly all of them shared a subtle flaw: in the louder passages, the sound of high pitched instruments--notably the violins--was tinged with a certain metallic hardness.'
>
> "A bit later on in his column, Mr. Fantel goes on to say, '....nothing the listener can do seems to smooth out or warm up the stone-cold sound of strings on digital discs. The mordant timbre seems caused by still unfathomed factors that have so far eluded conventional modes of analysis.' To all of which I politely but firmly say 'hog-wash' (substitute your own stronger language if you are so inclined).
>
> "So, what is it that's bothering listeners who are subjected to digitally mastered discs for the first time? Why do they find the treble tones to be overly bright? And, most importantly, why are they so quick to blame the new technology of digital mastering? In all probability, the fault lies with the record producer and/or the recording engineer.
>
> "To blame the new technology of digital recording for any sound aberrations introduced at the mixing console is ridiculous."

Any chance, Mr. Feldman, that "the certain metallic hardness"......"the stone cold sound of strings" on digital discs could be distortion caused by the sharp cutoff antialiasing filters?

Syn-Aud-Con is not impressed with the test conducted at AES in London "to determine whether or not the sharp cutoff filters......would in any way 'color' the sound of music being recorded digitally. The answer, to no one's great surprise, was that such filtering could not be detected even by experienced critical listeners."

"Hog-wash." Remember ads for the Edison phonograph? No one could tell the difference between the performing artist and the Edison phonograph. Is there no progress in listening tests since the turn of the century?

If interested in the subject, one may write the "AES Journal" and obtain a report, "Investigation Procedure to Obtain Perception Limits of Filter Related Sonic Degradations in Digital Audio Source Encoding," prepared by Korte, Bluthgen (of PolyGram) and Swientek, in which the statement is made in the abstract: "PCM source encoding techniques require mandatorily sophisticated low-pass filters for antialiasing and antiimaging purposes. Currently applied low-pass filters introduce sonic degradations like group delay distortion, pass-band ripple and ringing."

Or as Mr. Fantel of the *New York Times* says in lay language, "A certain metallic hardness"...."the stone cold sound of strings."

"The Mix," Volume 5, No. 3, has a splendid article written by John T. Mullin. John Mullin is the Army officer who had the good sense to ship home a German Magnetophon as a war souvenir. More importantly, he had obtained some of the high quality German tape (metal oxide) and had come across the high frequency bias modification for the Magnetophon and was sufficiently skilled to analyze it, duplicate it, and, finally, modify it--all in ways that retained its best older characteristics while improving it with his ideas.

John has entitled his article "The Start of Something Big" and, indeed, we haven't measured the magnitude of his accomplishment even at this late date.

John's step-by-step discussion of his interface with each key figure in pioneering professional tape recording is revealing of the ethical, gentlemanly manner in which he seems to approach all opportunities. His contacts with Col. Ranger, Jim Menard, Myron Stolaroff, Harold Lindsey and other early enthusiasts are of interest to anyone curious about how really important ideas are spread, grow, and come to fruition.

Harold Lindsey and John Mullin

How often the course first chosen is modified by circumstances. The creative man has several choices. Push pigheadedly on--if it works, its called persistence; choose new directions and benefit from the guidance of events until you emerge in a new era not predicted by the beginning direction. That's why geniuses are geniuses--they dare all on their best understanding of not always adequate data.

One example from the article will have to suffice as an illustration of what I'm saying here. A British officer told Mullin about hearing a Magnetophon that sounded far better than others.

> "Thinking this chap must have a tin ear, I bade him farewell and began to drive down the mountain. As my assistant and I reached a fork in the road, with all intentions of turning westward, I reconsidered. Suppose he had something there after all. So we turned eastward (towards Frankfurt)."

This decision led to the discovery of a Magnetophon with high frequency bias in place of DC bias which resulted in a remarkable improvement in dynamic range, lower distortion, etc.

It has been said, and wisely, that your decisions will master you. Be sure and read this very worthwhile bit of history. John not only tells what he did but, far more interesting to us, shares some of the choices that came up and reveals *how* he handled them.

Part I of "The Start of Something Big -- The History of the Tape Recorder."

TED KOWDRYSH recently sent us a "Resource Letter ENC-1: Environmental Noise Control" written by Thomas D. Rossing and published in the "Am. J. Phys." 46(5), May 1978. An eleven page collection of books, papers, and articles on all phases of acoustics of use in noise control studies. The writer's comments are, in my opinion, well taken and helpful in evaluating which of the mass of material deserves a reader's first attention. Our thanks to Ted for calling this collection to our attention. It reminded me of several books I can use and don't have.

SMILE

If you have ever had anyone take the wrong meaning from something you have said or written, you're ready for the following. First, there's that famous hymn, "Gladly the Cross I'd Bear" where someone thought that "Gladly" was the name of a cross-eyed bear.

BOOKS OF INTEREST

BOB BISHOP of Circle Industries in Harlingen, Texas, recently sent us an "ICS Electrical Engineers Handbook" printed in 1911. There is a full description of what Bob calls a "rock system volume control" on pages 198 and 199. It is an oil barrel rheostat and the remark is made "an ordinary barrel rheostat can carry a current of 90 to 100 amperes without boiling excessively."

Directions are included for the salt *solution* to be added to further vary the conductivity between the two metal plates at the top and bottom of the barrel. (Somehow, an oil barrel rheostat fits Texas.)

The testimonials in the rear of the book regarding the success of ICS graduates is eye opening. Engineers were considered "high pay" at $1500 per year.

We have always been impressed by the scholarship that went into these early correspondence school textbooks. This book's discussion of "wattless" current in a circuit with a low power factor (high reactance) is classic, showing how you get the detrimental heating effects of such current but no use of it in your load.

We have placed this book in our Syn-Aud-Con library with the appropriate notation indicating Bob Bishop as its donor.

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