SYNERGETIC AUDIO CONCEPTS

SYN AUD CON

newsletter

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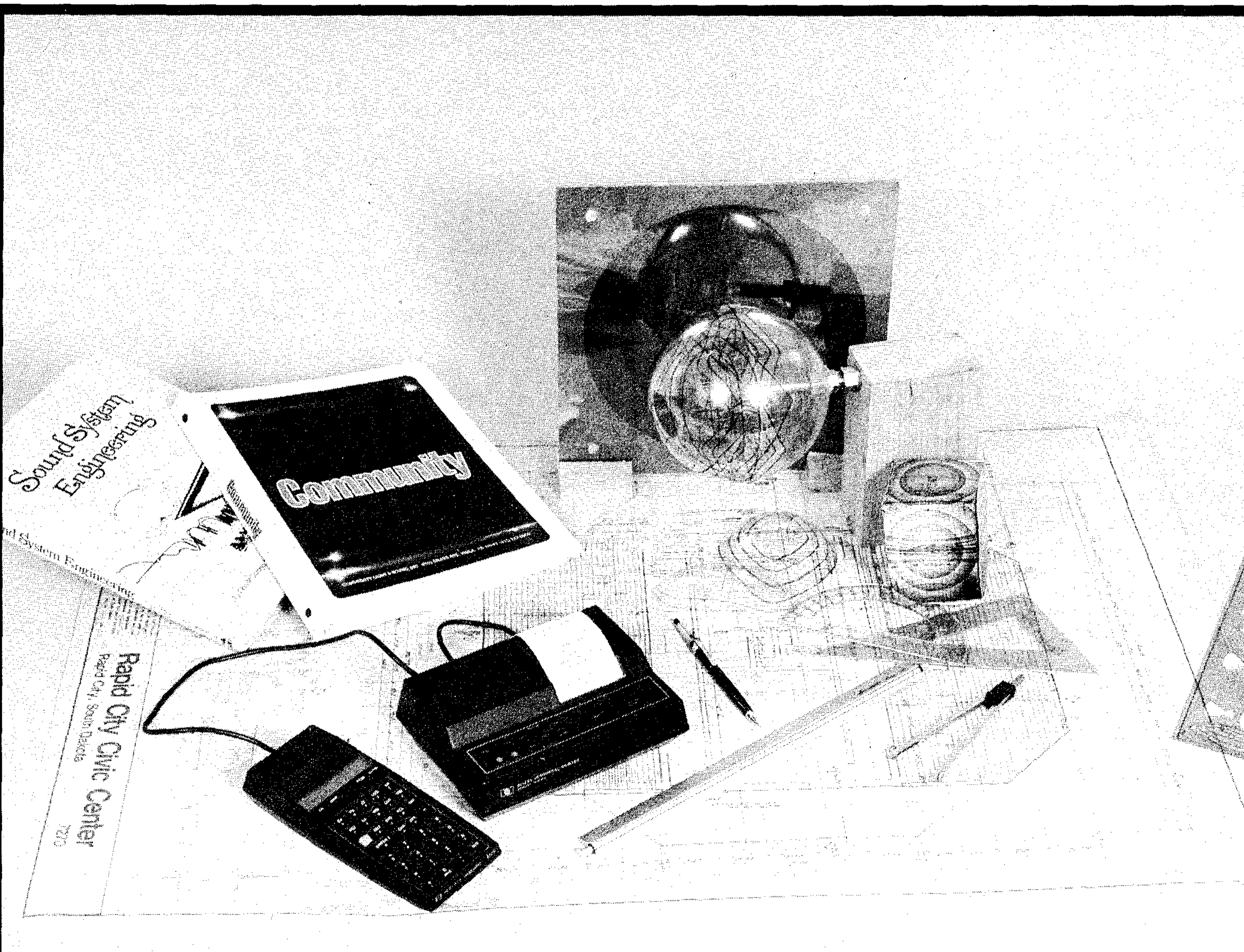
SUMMER 1982

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



SPHERICAL MAPPING FOR LOUDSPEAKER COVERAGE

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SYN-AUD-CON EDITORIAL

In traveling across this vast marvelous country of ours, we once again realized how great is the need for basic sound systems in the American lifestyle. Paging systems, background music systems, reinforcement systems, security systems, and many, many other forms of everyday communication are used in the communities we passed through.

Many of the small communities we passed through are not large enough to support a professional sound contractor but do require church systems, school systems, restaurant systems, etc. We can't help but believe that there exists as a real opportunity a way to train a local man in each of these communities to act as a representative of your firm's capabilities to each of the local potential sound system users.

Seminars for Church Laymen

Some of our graduates conduct their own seminars for potential customers such as church members charged with the responsibility of overseeing their church's sound system. (HOWARD PARKER of Sound Investments in California has been successfully conducting such seminars for about 5 years.) These seminars are at a level that informs the attendee in very basic terms of the fundamentals he must know:

1. What are the key components of a sound system (microphone, amplifier, loudspeaker).
2. How are they interconnected (hi Z, lo Z, balanced, unbalanced, etc.).
3. Choosing a microphone (omni, directional, high sensitivity, low sensitivity, etc.).
4. Choosing a loudspeaker (and, just as important, where to place it for most effective coverage with adequate acoustic gain).
5. Simple maintenance techniques (handling hum, noise, no-signal, and the basic substitution and isolation servicing techniques). Using today's very simple but extremely effective "signal tracers" can make this part of a layman's seminar very exciting.

These simple seminars should not include any mathematics, should avoid any theory not directly experienced by the participants, and should provide each attendee with "hands on" practice in walking over to a table with microphones, amplifiers, loudspeakers, cables and connectors and choosing from among them a system that he or she then hooks up and successfully operates.

Large contractors cannot profitably service small systems but smaller contractors can and should, whenever they can, get the customer to come to them rather than incurring the expense of going out to find them one at a time.

Send In Ideas

Syn-Aud-Con would like to encourage our readers to send in their ideas about how to advertise such meetings, what the course content should be, and what sort of tools they feel should be provided. We will publish the best of these in future issues and share the successes you have in exploring how to better serve the audio needs of our rural and small communities.

HIS SOUND - WORKSHEET

MIKE GARRISON of HIS Sound in Portland, Oregon, has developed a 6-page worksheet which is mailed out to churches over a wide area, many of them in small towns in remote areas. The worksheet, which is beautifully constructed to provide detailed information, is reproduced here (one page) to show the detail. We are asking permission of Mike to reproduce his worksheet in full as a future Tech Topic. We showed it to the Minneapolis class and they felt it was so important that DAVID WRIGHT (in the class) made photo copies for everyone.

Pulpit to front row seats ______
Pulpit to rear row seats ______
Width of front seating area ______
Width of rear seating area ______
Seating capacity ______
Are seats: pews ☐ theatre type ☐ chairs ☐

Do they have: cushioned seats ☐
cushioned backs ☐ no padding ☐
Are rows: straight ☐ angled ☐ curved ☐
(please illustrate on floor plan)

Pulpit to front row seats ______
Pulpit to rear row seats ______
Width of front seating area ______
Width of rear seating area ______
Seating capacity ______
Are seats: pews ☐ theatre type ☐ chairs ☐
Do they have: cushioned seats ☐
cushioned backs ☐ no padding ☐

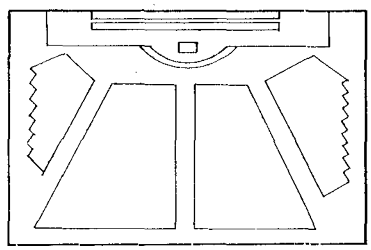
Are rows: straight ☐ angled ☐ curved ☐
Is seating area directly behind centered pulpit ☐
to one side of platform ☐ (left ____ or right ____),
facing inwardly ☐
(please illustrate on floor plan)

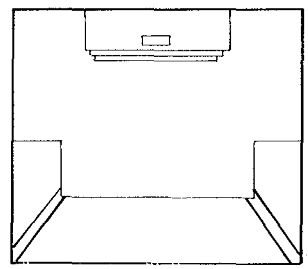
Depth of seating area ______
Width of front seating area ______
Width of rear seating area ______
Seating capacity ______
Are seats: pews ☐ theatre type ☐ chairs ☐

Do they have: cushioned seats ☐
cushioned backs ☐ no padding ☐
Are rows: straight ☐ angled ☐ curved ☐
(please illustrate on floor plan)

Depth of seating area ____/ ____
Width of front seating area ____/ ____
Width of rear seating area ____/ ____
Seating capacity ______
Are seats: pews ☐ theatre type ☐ chairs ☐

Do they have: cushioned seats ☐
cushioned backs ☐ no padding ☐
Are rows: straight ☐ angled ☐ curved ☐
(please illustrate on floor plan)



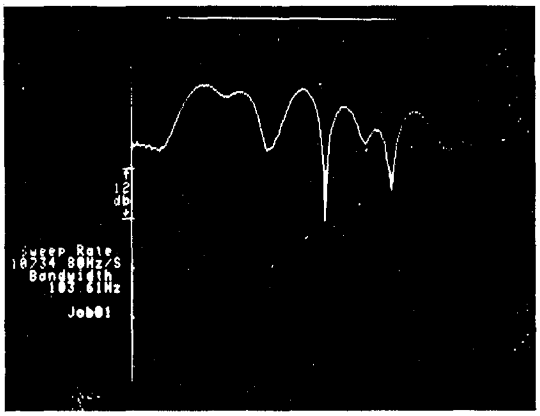


SMILE

According to LARRY ELLIOTT of Marshall Day Associates of New Zealand, they have solved how to deal with incumbent politicians. At one of their "Wild Life" animal parks, lions and other wild beasts run free. Visitors drive through in their automobiles to view the animals as they feed and frolic. A sign at the entrance cautions against opening windows, forbids convertibles, etc. At the end of a lengthy list, it concludes with "Politicians on bicycles admitted free of charge."

Larry Elliott, left, talking with George Owen, Chief Engineer at Rauland, and Harold Mosier, new General Manager at Emilar (back to us), during a break at the April Loudspeaker Array class.

A PROGRESS REPORT ON TEF™

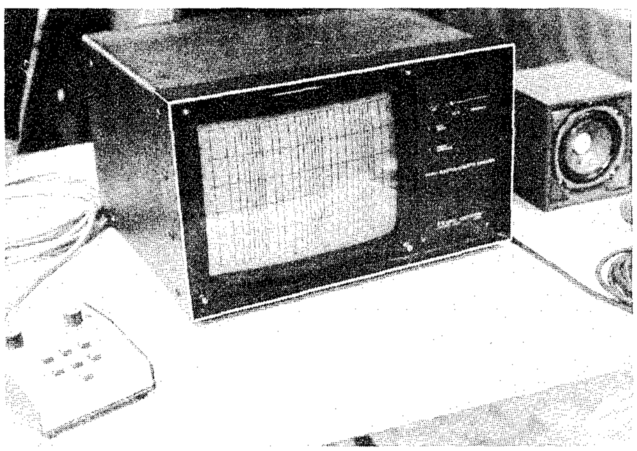


The photo shown has historic interest in that it is the first photograph of an Energy Frequency Curve (EFC) on the Crown TEF™ analyzer prototype. Seeing the prototype in a partial functioning state left us hopeful of the eventual outcome but aware of the tremendous man-hours of work remaining to be done. The display screen is of sufficient size to allow easy viewing. The potential versatility being planned into this analyzer will be a challenge to all of us to use effectively in the exploration of new measurement parameters.

MODEL QA-200 ELECTR-ACOUSTIC ANALYZER

During the San Francisco class, ED LONG and RON WICKERSHAM brought in and demonstrated the instrument that they make for quality control testing by the manufacturers using his loudspeaker designs.

Ed makes a careful distinction between "Quality Control" and "Quality Assurance." "Quality Assurance" is what you do to check incoming components made by suppliers for a manufacturer. "Quality Control" is the process employed to insure that your own manufacturing processes are producing the quality of product desired.



E. M. Long Associate's instrument for quality control testing of their loudspeaker designs.

Ron Wickersham, left, co-inventor of many product ideas that come from E. M. Long Associates. Ron owns Alembic.

Anyone building a quantity of loudspeaker drivers or loudspeaker systems would benefit enormously from owning one of these testers.

By merely pushing the five push buttons, tests such as:

1. Frequency response
2. Power response
3. Polarity
4. Loudspeaker resonance frequency
5. Impedance curves

are all performed automatically. The tester has exceptional accuracy and more than adequate resolution. The price, in the $6,000 range, is realistic and the performance is far better than we have encountered in this type of instrument in the past.

If you are in the market for a production line tester, we highly recommend that you contact Ed and Ron for further details as our description falls far short of all that this well-designed tool can do for a loudspeaker manufacturer.

E. M. LONG ASSOCIATES, 4107 Oakmore Road, Oakland, CA 94602. Phone: 415-531-8725.

A CONCERT HALL ACOUSTICS SEMINAR

For a very fortunate few able to afford such an experience, a ten year period of attending *live* concerts all over the world followed by an intensive apprenticeship to an outstanding acoustical consultant would be an ideal way to learn how to tell the difference between excellent, fair and poor concert halls. Even if you were this fortunate person there would still be genuine questions regarding the many halls heard because there would have been major differences in orchestras, conductors and even seasons of the year--affecting humidity, etc., in the concert halls.

One of Our Dreams

One of our dreams has always been to take the same orchestra, conductor and musical program to each concert hall of interest to us with as short an interval as possible between performances. If all our desires were to be fulfilled, an expert designer of world renown concert halls would accompany us to each of the halls and, after all the performances, *then* discuss what we *should* have heard and what it means in terms of design procedures.

We are now on the verge of accomplishing this treasured goal. We are planning to attend the European AES Convention in Eindhoven, The Netherlands (Holland), in March 1983.

Tentative Program

In addition to attending the convention, we want to spend three to four days to hear and see live concerts at different concert halls falling in three catagories of excellence. We want to arrange to hear the same orchestra, conductor and similar musical programs in all the halls. We will be accompanied by one of the world's most knowledgeable concert hall designers and members of his staff who will, on the fourth day, discuss in depth what we have seen and heard. The halls will range from superb to acceptable to poor. The orchestra and conductor will be a highly regarded one of first quality.

We will plan to fly from New York City to Amsterdam and then by private bus to Eindhoven. All accommodations, meals, transportation and concert tickets, both during the AES Convention and the concert hall tour, will be included in the price. We estimate a 14 day stay with one week devoted to AES and Concert Hall Acoustics Seminar. One week will be devoted to personal "touring."

We would suggest that each of the participants (limited to 30) acquire a copy of Leo Beranek's *MUSIC ACOUSTICS AND ARCHITECTURE at once* to use as a textbook in conjunction with the advance study assignments we will be sending to the fortunate 20-30 who choose to share this experience with us.

This offer is a tentative one subject to withdrawal at any time prior to submittal to you of a firm schedule and price. What we do request is that you let us know of your interest as soon as possible as the 20-30 participants will be selected in the order received by us.

We consider this a unique opportunity in concert hall acoustics as well as a chance to visit Europe under the most auspicious circumstances. It is expected that the Crown TEF™ machine will be along and that we will have the chance to demonstrate it in a number of spaces as well.

Eindhoven is the home of Philips and this trip offers the chance to visit this giant firm's "company town" in addition to all the other attractions present. We would be interested in knowing if you prefer the one week or two week schedule.

To sign up on the eligibility list, call Jan Kreitz at 800-854-6201.

SYN-AUD-CON HOSPITALITY SUITE AT AES

The AES Convention will start on Saturday, October 23rd, this year and will run for *5 days*. Syn-Aud-Con will exhibit as well as have a hospitality suite. Saturday will be devoted to exhibits only. Sunday will start with Sound Reinforcement related papers.

We don't know the entire schedule but the Calculator/Computer Workshop conducted by RUSS BERGER will be held on Monday. In addition to Russ, Syn-Aud-Con graduates TED UZZLE, MARK LAFFIN, JOHN LANPHERE and DON EGER will participate along with Deane Jensen. Don Eger is scheduled to introduce the TEF™ instrumentation from Crown!!!!!

We maintain the hospitality suite because we enjoy the relaxed time with Syn-Aud-Con graduates and maintain our exhibit to meet new people in audio. We appreciate it when grads can schedule time in the exhibit booth to help us.

Our suite will be in the Bonita Towers. You may want to check at our exhibit space to confirm the suite number to be sure no changes have been made. But, plan to be with us to meet old and new friends.

SYN-AUD-CON 1982 and 1983 SCHEDULE

We have planned Syn-Aud-Con Sound Engineering Seminars at our Seminar Center in San Juan Capistrano just before and just after the Fall AES Convention in Anaheim.

We are working on a series of Workshops for the rest of 1982 and for winter and spring of 1983. We will be sending you details on the Workshops in the next few weeks.

1982 Schedule

Chicago..................Sound Engineering Seminar............September 8-10
Washington, D.C..........Sound Engineering Seminar............September 15-17
Nashville................Sound Engineering Seminar............September 28-30
Orlando..................Sound Engineering Seminar............October 6-8
San Juan Capistrano......Sound Engineering Seminar............October 19-21
San Juan Capistrano......Sound Engineering Seminar............October 28-30
San Juan Capistrano......TEF™ Instrumentation Workshop........November 16-18
San Juan Capistrano......Microphone Workshop..................Nov. 30-Dec. 2

1983 Schedule

San Juan Capistrano......Sound Engineering Seminar............January 18-20
San Juan Capistrano......TEF™ Instrumentation Workshop........February 1-3
San Juan Capistrano......Installers and Grounding Workshop....February 15-17
The Netherlands..........Concert Hall Design Workshop.........March 11-13
San Juan Capistrano......Microphones & Teleconferencing.......April 5-7
San Juan Capistrano......Financial & Management Workshop......April 19-21
San Juan Capistrano......Loudspeaker Array Workshop...........May 3-5

There may be some shifting of Workshops and Seminars within the dates established. Anyone interested in the Workshops should let us know as we always notify those who have expressed an interest in a Workshop in advance of the general mailing. No money is required to put your name on the list for a particular Workshop.

NEWSLETTER UPDATE

We have prepared "Update" copies of the Newsletter. That is, important technical information from Newsletter Volume 1, No. 1 - Volume 4, No. 4, is in Update No. 1.

Update No. 2 is Volume 5, No. 1 - Volume 5, No. 4
Update No. 3 is Volume 6, No. 1 - Volume 6, No. 4
Update No. 4 is Volume 7, No. 1 - Volume 7, No. 4

Volumes No. 8 and No. 9 are still available individually.

Should you wish to obtain copies of these Updates, you may do so for $5.00 each plus $1.00 postage or $20.00 for the set plus $1.00 postage.

Index

We will have an index of Newsletters and Tech Topics available this summer. Full details will be available in the next Newsletter.

THE FAILS MANAGEMENT INSTITUTE

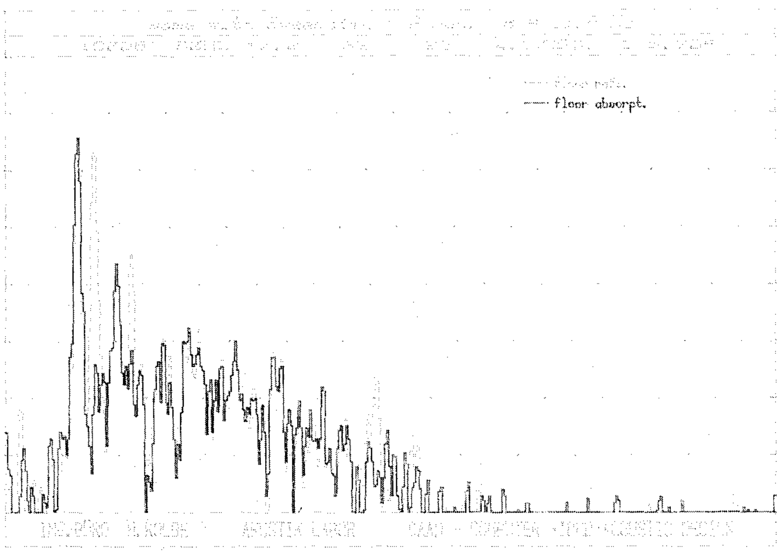
The Fails Management Institute points out to their newsletter readers that H.P. offers the following programs for H.P. 41C calculators: 5151 Glenwood Ave, Raleigh, NC 27612

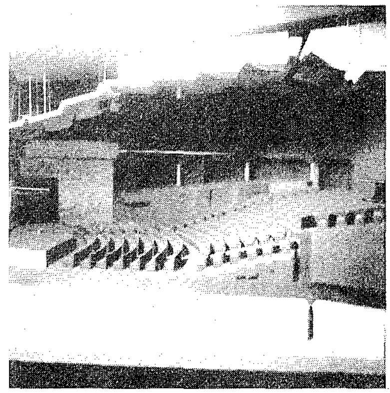
1. Business decision pack
2. Internal rate of return
3. Discounted cash flow
4. Annuities
5. Lease management
6. Depreciation
7. Bond analysis
8. Payroll analysis
9. Inventory analysis

They encourage the contractors they counsel to look into these programs and their adaptability to small contractor businesses.

TEF™ AND ACOUSTIC MODELS

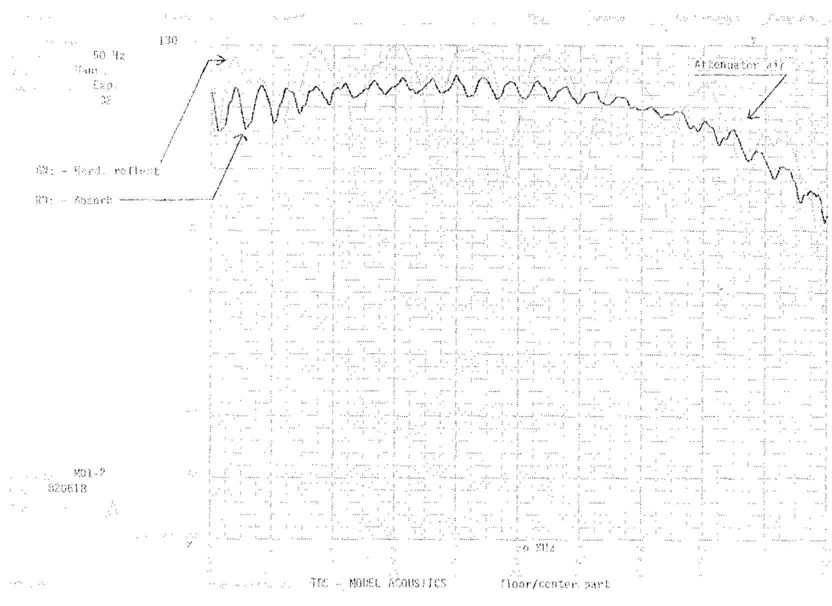
HELLMUTH KOLBE of Zurich, Switzerland, is easily the best equipped consultant in the world today in terms of modeling a difficult space and making TEF™ measurements of it. Hellmuth has had B & K





Transmitter MODEL Receiver
B & K 1/8" mikes

build him a special "black box" and special 1/8" transducers that allow him to engage in a 1/10th scale modeling technique for full range measurements or, with appropriate limits set on the upper frequency, models of 1/50th scale can be employed. (For ETC measurements full range is simply not mandatory.) Hellmuth has a complete B & K system in operation. We are particularly proud of Hellmuth because he went to the trouble to obtain a full TEF™ license through Syn-Aud-Con, take a special Heyser TEF™ workshop, and also go through a LEDE™ workshop in addition to a regular Syn-Aud-Con class. He did all of this, not because he was a beginner, but with an enviable background as an expert recording engineer and acoustic consultant.



Hellmuth combines remarkable experience with the recording arts, acoustical science, and the most advanced measurement tools and techniques in the world.

We find these measurements taken in a new concert hall *model* truly exciting and a perfect example of what the future will be in this category of work.

We hope to have Hellmuth help us conduct a workshop in the future on microphones and microphone placement, and his present work is rapidly providing him with insights that even his 3000-plus recordings in Europe over the past thirty years couldn't have done. Remember Hellmuth Kolbe's name--its bound to occur frequently and importantly in the future.

SMILE

The architect said to an acoustician, "I'll make the church so beautiful you won't care about the acoustics."

Anonymous

LEVELS AND TIME DIFFERENCES

During the Minneapolis class, we were able to demonstrate the highly audible effect caused by two loudspeakers at equal level but about 1" out of time alignment. (The effect sounds like a low pass filter is in the circuit.) At this point in the demonstration we tried mixing in a second demonstration. We switched in and out our 3 dB pad on one of the two loudspeakers. The results were startling to say the least. Over the majority of the listening area the subjective loudness *increased* by 10 dB when one of the two equal level loudspeakers was *dropped* by 3 dB.

(Newsletter Volume 9, No. 2, page 21, "Split Speakers? Reduce One Channel 3 dB," credits CLEON WELLS of Chicago Recording with asking during our last Chicago class, "What happens if?". Recently we were telling Cleon how pleased we are with his suggestion and he said the idea was first suggested to him by Mike King of Chicago Recording.)

In assembling combinations of loudspeakers, either all in one place or spread out over a wide area, it is important to be alert to which listeners will hear them at near equal levels but at differing time intervals.

This translates into concern whenever sound from two sources will arrive at equal levels but with time differences of 50 msec or greater (see the Doak and Bolt criteria, Fig. 7-46, page 130, of *SOUND SYSTEM ENGINEERING*) or with time differences of 1.0 msec or less. The area of from 3 msec up to 50 msec can, on occasion, be audible depending upon the number of accompanying reflections.

ALAN McCUNE of McCune Audio in San Francisco, which recently celebrated their 50th anniversary, tried many "what if" combinations during our May San Francisco class. It's during these "what happens if" sessions that we all learn so much - like reducing one channel by 3 dB.

J. Blauert and P. Laws Criteria

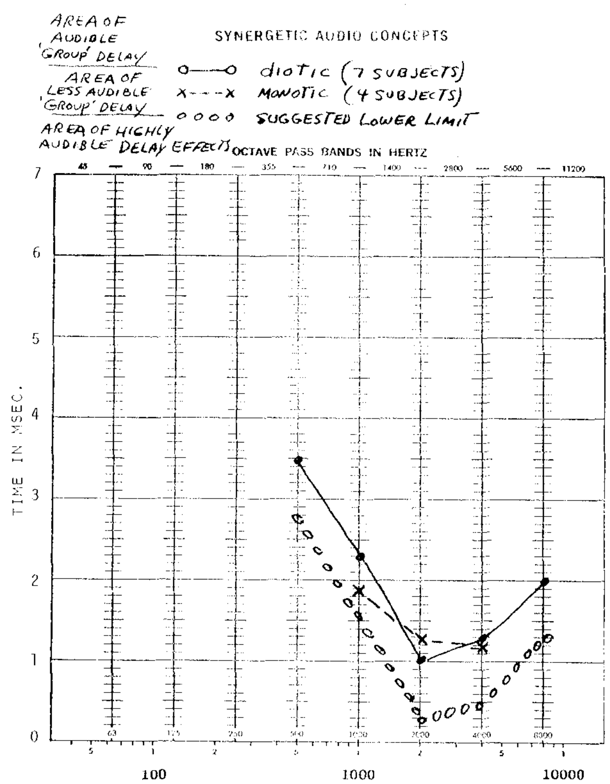


ILLUSTRATION #1

J. Blauert and P. Laws published a paper entitled "Group Delay Distortions in Electroacoustic Systems," J. Acoust. Soc. Am., Vol. 63, pp 1478-1483, May 1978.

This paper has had frequent referencing to it by many of the "literature researchers," most of whom do no listening--just reading.

Actual energy time curve (ETC) measurements accompanied by *trained* critical listening has repeatedly resulted in audible problems in two-way sound systems when the tweeter and woofer are within 1/2 inch of each other's *acoustic* centers.

We are beginning to suspect that the Blauert and Laws criteria are in reality the upper curve for an "envelope" whose lower limit represents the area of one inch or less of time difference. In other words, time differences of less than one inch are audible--from one inch to about two feet are not audible and then over two feet they once again become audible.

$$\frac{1 \text{ sec}}{1130 \text{ ft}} \cdot \frac{1000 \text{ msec}}{1 \text{ sec}} \cdot \frac{1 \text{ ft}}{12 \text{ in}} = \frac{\text{msec}}{\text{in}} =$$

.07375 msec or 73.75 usec

It is true that it is very difficult to hear the separation in time of two signals until it has increased to the length of time Blauert and Laws indicate. We merely feel that they may have overlooked the fact that a very small time separation falls into a special category at the bottom of their chart.

The Blauert and Laws criteria is correct in delineating 3 msec or greater as detectible but they fail to realize that their curve is the upper one for an envelope with a lower curve around 0.5 msec. (See Illust. #1)

Continued next page.....

TWO POSSIBLE MISALIGNMENTS

1.0" OF MISALIGNMENT (APPROX. 74 uSEC.) IS HIGHLY AUDIBLE. BOTH LOUDSPEAKERS RECEIVE EQUAL LEVEL SIGNALS (FULL RANGE SPEECH)

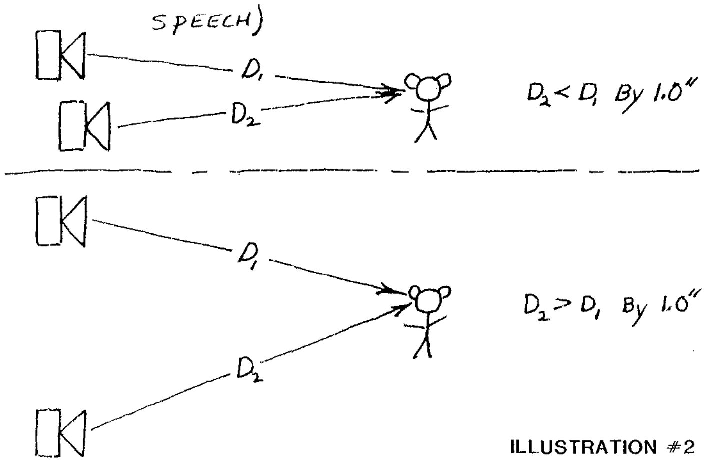


ILLUSTRATION #2

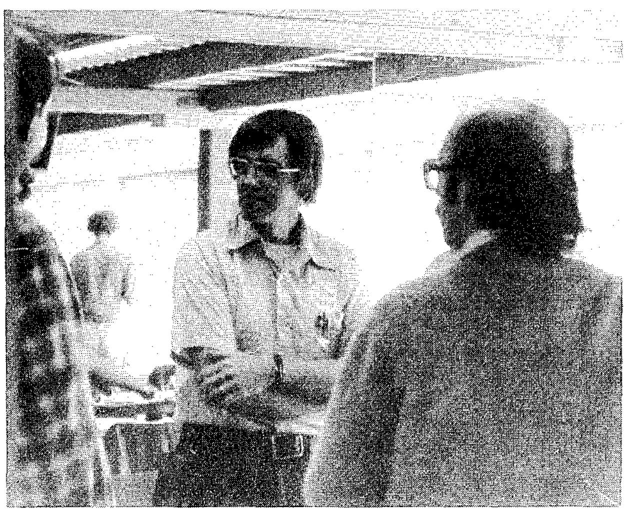
The Haas effect usually takes over for equal level signals separated by anywhere from 3 to 30 or even 40 msec. We can't help but wonder how many "dead spots" are not due to lack of coverage but to too much coverage from unequal distances actually canceling each other.

A Simple "Ear Opening" Experiment

Place two loudspeakers 1" apart and feed them equal level signals. (See Illust. #2) Sit about ten feet away and listen. Have someone switch first one on and then both on. Move through the sound field and hear how broad the area is that is affected by this effect. Then with both units on, *reduce one* of them by 3 dB. Note the increase in both high frequency response and, but even more apparent, the rise in loudness.

Now place the two loudspeakers 5 feet apart and repeat each part of the experiment. We believe that you will quickly agree with us that the bottom limit that should have been included in the Blauert and Laws chart is by far the most audible.

PHASE DISTORTION & PHASE EQUALIZATION



Don Keele during a break at the Loudspeaker Array class in April.

DON KEELE called our attention to a very thorough and fundamental tutorial review entitled "Phase Distortion and Phase Equalization in Audio Signal Processing" by Douglas Preis (presented as a preprint at the October 1981 AES Convention in New York City). It includes an excellent discussion of Wigner plots and other suggested "mapping" schemes.

Farrel Becker, left, discussing a FTC plot with Glenn Meeks.

Reading this paper and the recent Philips paper leaves one with immense respect for FARREL BECKER's pioneer FTC plots first shown in the Syn-Aud-Con Newsletter of the summer of 1980 (Vol. 7-#4, page 21). GLENN MEEKS of Indianapolis started making FTC plots about the same time as Farrel but Glenn's work was not published until later. Mr. Preis is probably genuinely unaware of Becker and Meeks' work because he fails to list their papers in the bibliography but there is absolutely no question in my mind that the "ambience" created by the discussion of their work around earlier AES Conventions was the catalyst that triggered many an academic's subliminal "knee jerk."

Preis' preprint is still available from the AES (1849 (J-2) 1981 Oct. 30 - Nov. 2 Convention).

C. A. "Puddie" RODGERS

PUDDIE RODGERS passed on early this May in Seattle, Washington, after a lengthy illness. The audio industry lost one of its most promising young people. Her remarkable paper, "Pinna Transformations and Sound Reproduction," in the April 1981 AES Journal and her Ph.D thesis remain as monuments to her intellectual victories. To those of us in Syn-Aud-Con fortunate enough to have known her and shared ideas with her, we treasure the memory of a woman dedicated to her work and loyal and warm to her friends.

C. A. "Puddie" Rodgers at the Heyser TDS Workshop.

Scholarship Fund

Syn-Aud-Con is proposing a "C. A. "Puddie" Rodgers Fund" whose principal will be administered by a suitable Board of Trustees and whose income will be used to provide financial awards to "women capable of making significant contributions to the field of acoustics."

It is our belief that Puddie would be pleased could she know of this and that it is a fitting memorial to provide tangible assistance to other women like Puddie.

THE INITIAL TIME DELAY GAP

Beranek, in his classic work, *MUSIC ACOUSTICS AND ARCHITECTURE*, identified the initial time delay gap (ITD) as a key parameter in the acoustic quality of a concert hall. The ITD is defined as the time in milliseconds (msec) between the arrival of the direct sound at a listener and the arrival of the first *significant* reflection. A reflection's significance is dependent upon its level in decibels (dB) as compared to surrounding scatter and its time interval. At the Syn-Aud-Con Seminar Center, the *liveness* of our high ceiling area is judged to be greater than the *liveness* under our low ceiling area. Consequently, most observers are surprised to find that the reverberation time (RT_{60}) for both areas is the same. There is a true reverberant sound field to measure in this large 80' x 60' room with its 30 foot ceiling.

This was of great interest to us because it lead our Energy Time Curve (ETC) investigations into a new direction in relation to its use in large reverberant rooms. The sense of *liveness* obviously depends far more on the ITD than on the RT_{60}.

Until the advent of the TEF™ measurement system, there was no reasonably accessible way to acquire such measurements. There were those who equated single frequency impulse data with ITD significance but the spectral energy distribution in time was a relatively unknown area. It required the combination of total spectral energy, along with a dynamic range of at least 60db, to view the presence or lack of significance in the energy's temporal distribution.

Distinction Between ITD and Immediate Reflections

There are time delays within the first few msec (a path difference of say less than four feet) that cause significant spatial misdirection to a listener via the comb filters thus generated but that are more than likely too short to be considered the ITD. Just how short a time interval can qualify as an ITD rather than as a comb filter generator is not delineated in the technical literature but I would suggest as a first approximation that any time interval too short to create a partial Haäs effect probably would serve as a line of demarcation between the ITD (long enough to create a partial Haäs effect) and the comb filter interval (CFI) (short enough to cause significant comb filters but not long enough to provide even a partial Haäs effect). According to Madsen's work (see fig. 6 in Tech topic Vol 8 #2) the onset of the Haäs effect can be as early as a few milliseconds, especially in very dead rooms, or in typical meeting rooms, out to 20 or more msecs. This is an area wherein fruitful original work remains to be done.

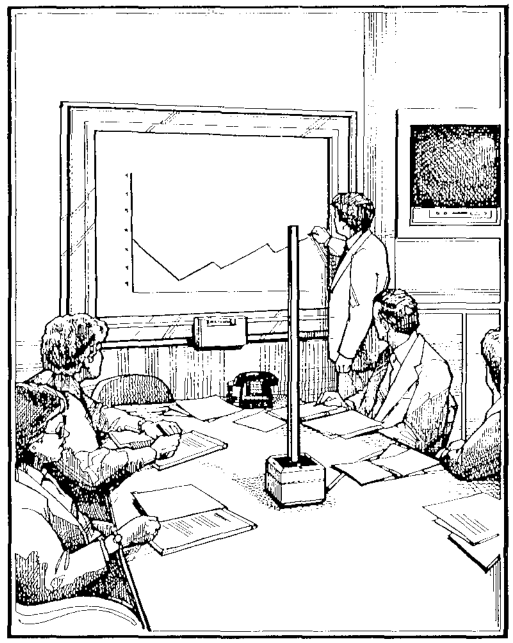
Rates of Decay vs RT_{60}

Superficially, the rate of decay (RD) in dB/sec can be considered to be $RD = \frac{60}{RT_{60}}$

Actually, in rooms as small and absorptive as control rooms and probably a majority of studios, there is no RT_{60} to measure (the reverberant sound field is *below* the ambient noise level (L_{AMB}). There is, however, the rate of decay of the first three or four reflections in each axis of the space. Often these RDs are radically different for the different axes. Should they be so or would uniformity be a virtue? Again, an area for fruitful original work has been identified. What effect does this RD have on the establishment of a reflection's significance as an ITD? Is even spacing in time better than uneven spacing? My guess would be that uneven spacing would imitate nature the best. For the first time in the total history of building recording environments relevant measurement techniques are available. The challenge is to ask relevant questions and prepare measurements that address themselves to the psychoacoustics inherent in small volume non-reverberant spaces wherein the early reflected field is dominant.

"QUORUM" TELECONFERENCING MICROPHONE

Bell Laboratories, Holmdel, New Jersey, has developed an "array microphone" that utilizes 28 individual electret transducers mounted along one side of a 30 inch stalk. Claimed benefits include less than 1/r attenuation (i.e., less than 3 dB per doubling of distance) and bipolar directionality including a null above and below the array. Very small transducers, antenna theory, and careful application of Chebyshev filters result in, we are told, a surprisingly wide bandwidth over which significant directional control is exerted. Those wishing to test such microphones can "lease" them from the Bell operating companies for a tariff rate of approximately $60 per month.



NEW "HI-FI FAD?" - THE LEDE™ LISTENING ROOM

Much to our pleasant surprise, our Live End-Dead End™ (LEDE) is being recognized by many critical home listeners. Among the professionals in the recording industry, many that we respect have concurred that LEDE is all its claimed to be and even a little more. We have been receiving startling confirmations of LEDE superiority from sources both here and abroad that make clear the eventual recognition of the concept by reasoning engineers.

How pleased we are to hear from skilled music critics that an LEDE home listening room also provides easily discernible improvements in terms of their listening environment at home. Bert Whyte has written in July Audio magazine, "I can truthfully say that my LEDE room has given me more unalloyed listening pleasure than I ever expected to hear in my home."

> "....those pesky close-order early reflections normally produced in all non-LEDE listening rooms are absorbed. Listening to a familiar loudspeaker in the LEDE environment, one is immediately aware of the great increase in the clarity and cleanness of the sound. Bass is singularly free of boominess or overhang. Instruments seem more clearly defined. Dense and complex musical textures are more articulate. Depth perception is greatly enhanced, as is the sense of ambience and the reverberant characteristics of the recording hall. Loudspeakers with superb imaging, like the B & W 801F, achieve new heights of instrumental localization and positioning; speakers with lesser degrees of imaging are considerably improved."

Mr. Whyte goes on to discuss the total cost of treating his living room with the required materials (including Sonex), "The costs came to $1,056.00 for Sonex and a $5.00 rental of the stapler. Not cheap, I'll agree, *but I can't think of any comparably priced audio equipment that would even approach the dramatically heightened sense of realism afforded by Sonex in the LEDE configuration.*" (Italics mine)

Mr. Whyte also touches directly on why some contemporary recording engineers are uncomfortable with the LEDE concept.

> "The LEDE configuration allows you to get further into the music, be it performed in a studio or concert hall. However, therein lurk some annoyances. What once was masked by the interaction of listening room acoustics with loudspeakers is now all too clearly revealed. Poor mike techniques, especially multi-mike, can be devastating to hear--for example, some gross "pan-potting" in which you can plainly hear left/right manipulations and "gain riding." In addition to all kinds of sonic anomalies, there are matters of performance. Poor intonation becomes easily apparent, as does inaccurate "unison" playing."

All of the above is indeed true and Mr. Whyte's two articles in Audio (the first in June and the second in July) contain helpful details for the "do it yourself" home experimenters.

In one survey of sixty control rooms, the premier LEDE room used as a reference was never exceeded and, indeed, we are told was only approached by one other room.

When we cast LEDE upon the waters, we felt that such basic acoustic truths did not need "hype" to achieve eventual meaningful acceptance. Thus, we are heartened by the increasing numbers of individuals who are reinforcing our belief that "better ways do prevail."

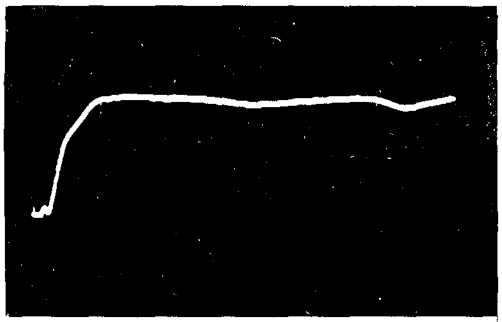
THE "REVOLUTIONARY" COPERNICUS SPEAKER

The small truncated globe speaker that has intrigued us for sometime is now being marketed by J. W. Davis and Company of Dallas, Texas. We would like to see it called the "Copernicus."

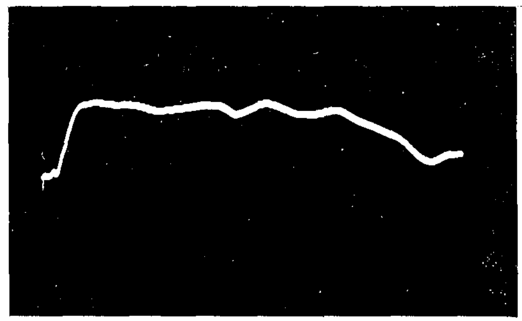
Nicolaus Copernicus was a distinguished churchman and a humanist intellectual from Poland. Born in 1473 he, by 1543, had finally braced himself to publish his mathematical description of the heavens, what he called "De Revolutionibus Orbium Coelestium"--"the Revolution of the Heavenly Orbs" as a single system moving around the sun. (The word "revolution" has an overtone now which is not astronomical, and that is not an accident. A revolutionary was one who, with Copernicus, came to believe that the earth revolved.) Copernicus had his first glimpse of his manuscript on his deathbed.

Copernicus' theory--and it was only that at the time--waited almost 200 years until Newton finally proved that the earth indeed is in motion.

We truly feel that "The Copernicus" with its unique truncated spherical housing reflects the tradition of its namesake.



1,000 - 10,000 display of a 0 to 10,000 sweep.



1,000 - 20,000 display of a 0 to 20,000 sweep.

It's a theory that works but still awaits a full explanation of why. It's causing a revolution in thought regarding small systems, and we believe you'll see many of them in orbit in the immediate future.

We have reproduced TDS frequency response curves, which shows one reason why it sounds so good.

Is the "Copernicus" for you? A revolutionary new way to overcome "flat earth" audio.

FROM THE LONDON FINANCIAL TIMES

Mahlon Burkhard, Director of Engineering at Industrial Research Products, who has given the industry exceptional digital time delay units and the automatic mixer, shared with us the following:

> *Disputed problems on the frontiers of knowledge cannot be decided by majority vote. Otherwise Galileo would have had to accept the verdict of his peers that the sun moves round the earth; and cosmonauts would not have ventured forth today for fear of falling off the edge of the universe.*

DIGITAL ADDITION CIRCUIT

Believe it or not, someone has patented a decibel addition circuit (digital) that performs the familiar

$$10 \text{ LOG } (1 + 10^{d/10})$$

Where: 1 represents the larger value of two values

d represents the difference in dB between two levels

See JASA, page 1063, April 1982.

4,290,111

43.85.Fm DECIBEL ADDITION CIRCUIT

Keith Dillon, assignor to The Singer Company
15 September 1981 (Class 364/768); filed 12 December 1979

This digital circuit for obtaining the level of the combination of input levels L_1 and L_2 (which is larger than L_1), functions in the familiar manner of calculating the difference $d = L_2 - L_1$, looking up $10 \log(1 + 10^{d/10})$ in a read-only memory, and adding this logarithm-addition function to L_2. --RWY

APRIL 1982 CLASS

QUESTIONS REGARDING SAMPLING RATE IN DIGITAL AUDIO

The following is an excerpt from a letter and an article from PAT QUILTER of QSC Audio Products in Costa Mesa, California, covering questions regarding the sampling rate in digital audio:

I am enclosing a copy of a letter which I wrote some months ago regarding the "digital controversy." I share your determination to expose this new technology to the most stringent and rigorous criticism, before we commit our audio treasures to this format. This, of course, is the heart of the scientific method. I had hoped to be able to perform some quick "hands-on" tests of the popular digital recording machines, but being the sole design engineer for a small business leaves me with no time to arrange this. Consequently, I have no actual data to support or refute the questions raised in my letter. I have asked every experimenter in sampled audio (who I meet at AES shows, etc.) if they have experienced the "second or third-order alias distortion" which I am concerned about. Out of perhaps 15 queries, about half have said "Oh, yes, definitely" and the remainder have never noted any such effect. The people who have confirmed the problem were mostly actual designers of sampled systems; those who have not noted the problem were generally users.

In reviewing my original paper, which has been set aside on my desk for some time, I still stand by the introductory comments and (admittedly crude) experimental results; however, I can expand here upon several points which I did not make very clear.

1. A simple thought experiment. Let us assume that a pure sine wave is sampled exactly at the Nyquist frequency. This waveform could be sampled anywhere from the crests to the zero crossing point. The resulting output can therefore vary from zero to a level equal to the sine-wave peaks, depending on the phase relationship between the input frequency and the sampling clock. Now, let us move to a frequency slightly below the Nyquist frequency. The sample-and-hold output will now vary from zero to maximum at a rate equal to the difference between the Nyquist frequency (half the sampling clock rate) and the input frequency. If, for instance, we are 100 Hz below the Nyquist frequency, we will have a 100-Hz "beat" effect in the output. As we continue to move lower in frequency, the "beat" increases in frequency until it moves outside of the audio range. This implies that there is a term in the output equivalent to F (clock) - 2F (in). Right away, we have demonstrated alias distortion at frequencies significantly below the Nyquist frequency, but if the problem were limited to this, it would not be too serious. However, I suspect that there is a diminishing series of "IM products" of the form F (clock) - nF (in). My crude sampling test revealed that the IM products were significant for "n" as high as 6 or 7. If true, this clearly implies that sample rates on the order of 50 KHz can impair audio frequencies as low as 6-8 KHz, which can occur at high enough levels to be significant. The resultant IM products will, indeed, be harsh, dissonant, and may occur at low frequencies which might trigger instinctive "avalanche" or "pre-earthquake" stress responses.

2. At no time in any of the literature I read, have I ever seen a distortion vs. frequency graph of the proposed digital recording systems, or any performance claim beyond the usual unqualified ".03%" or so, based on the digitizing error, at full scale.

Anyway, I apologize for dumping these essentially half-baked questions and assumptions on you, and I wish I had the time to follow up on these questions. Rather than let the issue go down the drain, I would rather at least offer the ideas to someone with greater experience, more contacts among those who would know some answers, and with a clear interest in the subject. I would greatly enjoy any light which you can shed--it would be a relief to find out I'm all wet, and if the questions are productive, perhaps it will lead to an improved system, such as a higher-sampling-rate delta-mod of some sort.

(Signed) Patrick H. Quilter

Questions Regarding the Sampling Rate in Digital Audio

Article by Patrick H. Quilter (4-3-81)

There has been much concern, as with any new technology, over the audio performance of digitally encoded sound. In this case, the concern is especially well taken, as unlike the effects of reproduction hardware or specific analog recording techniques, the effects of digitizing will pervade the entire body of recorded source material, once the industry has completed the change-over. Whatever digital standards are selected will place inherent mathmatical limits on possible performance, so it is obviously important that the industry not "build in" limitations which future generations will regret.

In reviewing the available comments, it appears that most people are impressed with the superior low-frequency linearity and low noise floor of present digital audio. However, there is a persistent body of criticism regarding "harsh" or "sour" high frequency reproduction; one researcher (Dr. Diamond) even claims to identify stress in his subjects, caused by listening to digitally recorded sound. To what can we ascribe these effects?

Barring some completely unknown physical phenomenon associated with the playing of digitally mastered records, we virtually have to assume that there is something happening within the audio passband. As far as I know, the listening tests and "digital stress" demonstrations are performed using readily

Continued next page.....

available stereo equipment and LP records; one can hardly expect such gear to consistently emit significant energy outside of normal audio frequencies just because of playing digitally mastered recordings. It might be possible that residual ultrasonics at the sampling rate are present, but one would expect at least some of the sources to have filtered these out. In addition, people are exposed every day to much higher energy ultrasonics, such as burglar alarms and TV scanning harmonics, without any medical problems. Thus, I would be surprised if the consistency of observations could be caused by any low-level, inconsistent residual.

Much attention has been given to possible bad effects of the anti-alias input filter. Considering the repeated studies over the years of the limits of human hearing, and the low sensitivity of the ear to phase shifts, it comes as no surprise that sharp cutoffs at 16 KHz are barely audible, and at 20 KHz become inaudible, or at least not detrimental even if observed. Therefore, this part of the system is apparently acceptable. Likewise, there is ample evidence that 14-16 bits provides sufficient resolution for low distortion and wide dynamic range.

So now we come to the perhaps most controversial variable--the sampling rate, specifically as it relates to the sample-and-hold process which precedes the actual analog-to-digital conversion. This process can be viewed, in a sense, as the non-linear mixing of the input signal and the sampling frequency. It is well-known that such a process creates intermodulation frequencies which occur at the sum and difference of the two original frequencies. In fact, Nyquist's famous theorem states that for input frequencies greater than half of the sampling rate, there will be "alias frequencies" which appear below the input frequency (i.e., within the audio passband). However, not much attention is paid to the additional IM products which appear at sum and difference frequencies of multiples of the original frequencies. These secondary, tertiary, etc., alias frequencies can be generated by input frequencies well below the Nyquist limit. The resultant distortion would be most objectionable, as new frequencies would be created with non-harmonic, arbitrary relationships to the original input tones. Familiar examples of this type distortion would be the ring modulator effect, often heard on off-frequency single-sideband transmissions. However, the impression I get from the literature, and from talking to digital sound engineers at the AES shows, is that one is "home free" below the Nyquist limit.

The question finally stimulated me to take some time from my regular work to perform a simple experiment. This consisted of cobbling up an elementary sample-and-hold circuit, with a swept sine wave input. I monitored the results through an amplifier and speaker, to determine at which point any alias distortion would become obvious. My experiment was crude, but the results raise some very important and fundamental questions which the industry must address before finalizing the sampling rate.

Here are my results: I used a sample rate of 40 KHz, and varied the input frequency from 2 KHz on up. As expected, the region around the Nyquist frequency, 20 KHz, demonstrated clearly audible beat frequencies which started from 0 Hz at exactly 20 KHz, and which increased in frequency as the input was raised above 20 KHz. This, of course, demonstrates the expected response for inputs above the Nyquist limit. However, the same sequence of rising frequency was observed as the input was decreased below 20 KHz. As the input frequency was further lowered, a new, somewhat lower-level "zero-beat" null was observed at 13.3 KHz, another at 10 KHz, at 8 KHz, and just barely audible at 6.67 KHz. In between these points, I observed alias frequencies climbing away from each null. In short, I observed a diminishing series of secondary, tertiary, etc., alias distortions for frequencies well below the Nyquist limit. With my casual set-up, the distortion became inaudible below 1/6 of the sampling rate.

One secondary effect which I examined was the aperture time of the sample-and-hold circuit. Varying this from 5% to 50% did not affect the amount of distortion, leading me to conclude that this type of IM distortion is, indeed, a basic attribute of the sample-and-hold process. I also used the low-distortion sine-wave oscillator on our Sound Technology 1700B (.002%) to verify that the alias products were not caused by harmonics of the input signal.

If my conclusion is correct, and unless the digital recording manufacturers are using some undisclosed, and very sophisticated pre-conditioning of the input signal, the ability of present-day digital systems having 44 KHz or 50 KHz sampling rates to cleanly reproduce audio signals in the 5 KHz - 20 KHz range is open to serious doubt. I submit, in fact, that the presence of significant IM products from these frequencies is causing the audio objections and stressful listening responses.

The biggest question in my mind is why such an obvious problem has not been discussed in the papers pertaining to digital audio. The equipment designers must surely be aware of this phenomenon. My goal in reporting this experiment is to encourage a few simple tests, using the present recorders, which will quickly prove or disprove the validity of these objections. The first test is simply to apply a sine wave input, and sweep it up to the limit of the alias filter. If clearly audible "birdies" are not heard from about 5-7 KHz on up, then the designers may be congratulated for solving a basic problem. On the other hand, especially if distortion is heard, I would look for further measurements, leading to a distortion-vs-frequency curve. I will be very much surprised if the high frequency performance is not far below what is now routinely demanded of power amplifiers and other recording/playback electronics. With such a curve, then accepted standards of tolerable IM distortion can be applied to dictate a sampling rate which future generations of music lovers can live with.

PZM® EXPERIMENTATION

One phonograph record to keep on the lookout for will be the excerpts from the winners of the Crown PZM® contest. We are told that they intend to release samples from the winning recordings. Here is an example of the kind of truly innovative microphone work one of the winners, MIKE LAMM of Dove & Note Recording in Houston, Texas, utilizes in making his recordings. While a majority of conventional microphone engineers continue to sit on their "mental hands," the real talent is accepting the challenge of getting on with necessary hypothesizing, experimentation and refinement of PZM® technology.

Last September in the Denver class the idea occurred to me to use two PZM^2 in an O.R.T.F. spacing (Diagram #1). Further thought revealed that by approximating a cardioid pattern with the plates angled at 120°, the required O.R.T.F. angle of 110° could be achieved also.

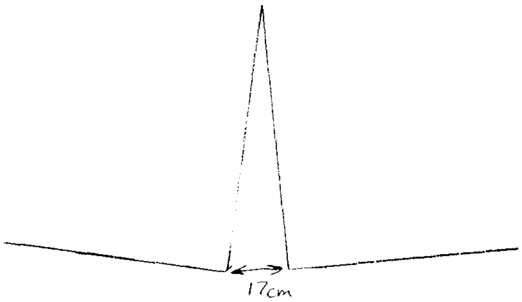


Diagram #1

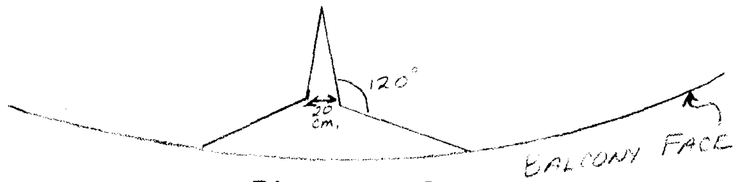
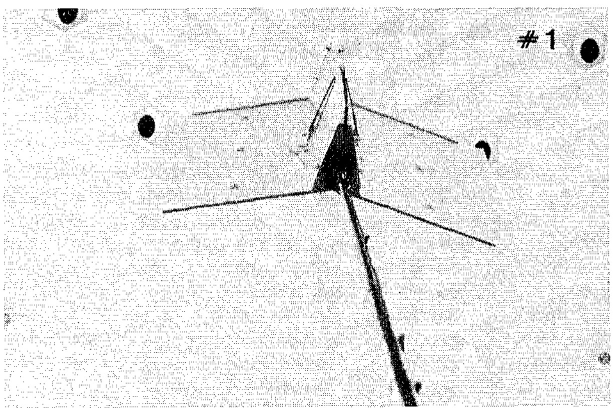
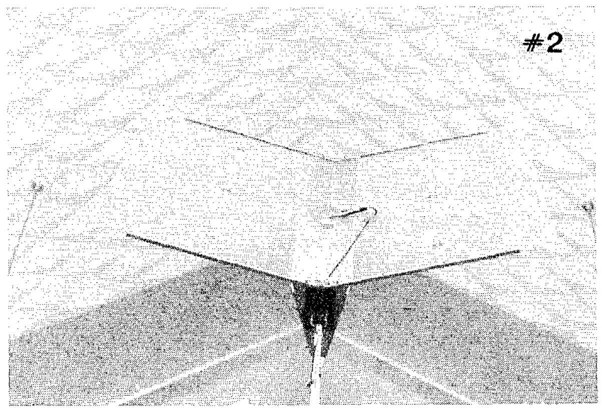


Diagram #2

The first prototype (a D.I.N. type) was built in early April and was used twice. The first time, I hung the array from the front of a balcony (Diagram #2). I did not consider this to be optimum placement because of possible early reflections from the balcony front. Also, field experience showed that fifty pounds was an unwieldy weight to try to handle for temporary set ups. However, the recording turned out well and won an Honorable Mention in the "PZM Challenge." I was convinced that I had to build a lighter weight model and a sturdy stand to support it.





The second prototype weighed 13 pounds and was a D.I.N. configuration. I first used it at a choral festival held at a large church in Galveston. Picture #1 shows your PZM 150, Model BP, serial numbers 0102 and 0103 mounted on the front of the array. Picture #2 shows the back side of the array with a Crown model 31S mounted to pick up hall ambience. The array is attached to the stand via a modified Omnimount model 100WA. This ball and socket mounting allows great flexibility in mounting angle combined with high strength.

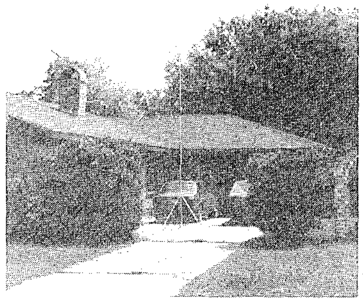
The array was next used to record an all-male chorus at South Main Baptist Church, Houston. Picture #3 shows the stand *almost* fully extended. Incidentally, when I took that picture there was a 10 mph breeze blowing! As you can see, the stand is very stable. Picture #4 shows the third prototype (an O.R.T.F.) set up at my own church where I first began recording five years ago. The balcony choir loft railing is twenty-four feet above the sanctuary floor, and it was in this location that I first started thinking about a tall microphone stand several years ago. The bottom of the array is about 30 feet above the floor. This third prototype, besides being an O.R.T.F. configuration, has an aluminum channel frame epoxied around the edge for support. There are three hinges on each rear "wing" so that the whole assembly folds up into a package 4 in. high by 48 inches long by 18 inches wide. It can be assembled in less than 15 minutes by one person and can be changed from O.R.T.F. to D.I.N. or N.O.S. configurations simply by substituting the required top and bottom plates (also aluminum). See Picture #5. Picture #6 is a set up for a junior high school band concert.

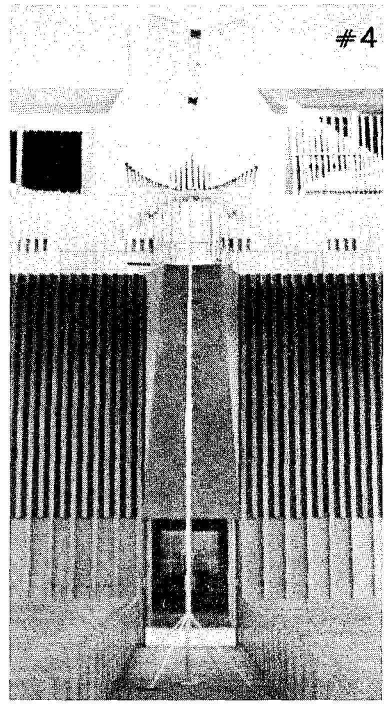
Experience this spring indicates that the aluminum stand will safely support speakers to about 20 feet, PZM arrays to 30-32 feet, and microphones on 5/8" - 27 thread mounts to the full forty feet! It breaks down into two sections, each 4 feet long, weighs 23 pounds, and is infinitely adjustable in height. Using four legs for greater stability than the usual tripod, each leg is independently adjustable for sloped or uneven floors. I am in the process of replacing the chain with a rigid strut member for even

Continued next page.....

greater stability and infinite independent adjustment of each leg. I am also taking your advice about eliminating the need for tools wherever possible and am looking into the matte gray or black finish to reduce reflection of light off the stand. I don't know of any other stand that is as adjustable, as light, as stable, as versatile (speakers, PZM array, or standard microphones), or as tall. Can you tell that I'm damned proud of it?! It *is* expensive and that may keep it from being marketable. I only hope I can sell a few to get back my development costs. I'm afraid that between material costs, machine shop costs, anodizing, and my labor I'd have to charge at least $500 unless quantities were such that my costs went down. I was originally aiming at a selling price competitive with the Matthews stand but that just hasn't proved possible. I *do* think it's a better, more flexible design and the only stand I'd trust to support a 4' x 4' plate or PZM array. I'll admit to bias on the subject, too.

#3





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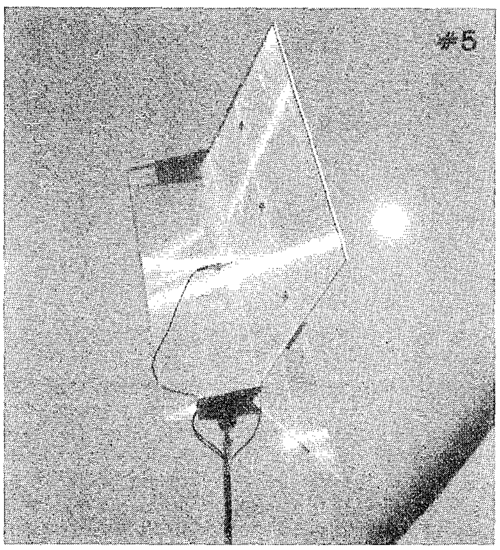
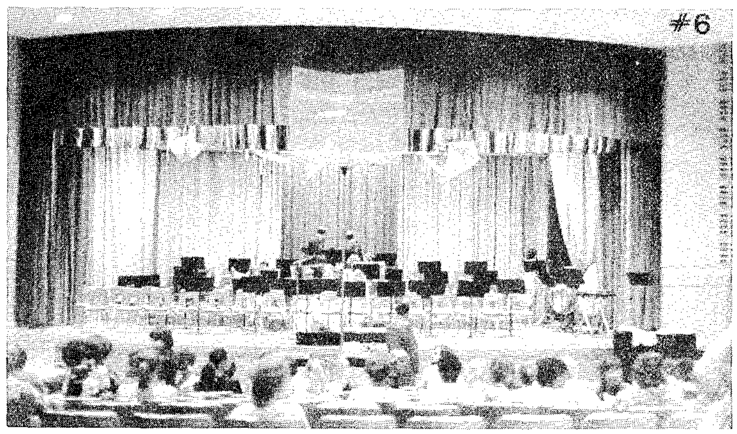


#5



#6

The weight reduction on the array with no reduction in size was possible by using the hollow wall material enclosed. It is polycarbonate, easy to cut, weighs 0.25 lbs./sq.ft., and is 30% less expensive than 1/4" plexiglass. Its only drawback is that it is not totally transparent due to the internal ribs. The arrays I've built with it have been both light and strong, and have the Omnimount socket on the bottom for stand mounting and lifting eyes on the top for flying the array with nylon cord or heavy monofilament.

Theoretically, I see the following advantages and disadvantages to near coincident PZM arrays:

1. Both time and amplitude information in the recording.
2. No off-axis coloration.
3. Good audience rejection in live recording.
4. A high degree of control over ambience by use of the third (rear) mic.
5. New materials with better size-to-weight ratios make possible large arrays (with good bass response) that are lighter and therefore more practical than ever before.
6. With the high stand or the lifting eyes, placement is almost unlimited.

If you are interested in communicating with Mike Lamm, his address is:

Dove and Note Recording Company, 15415 West Antone Circle, Houston, TX 77071. Phone: 713/723-7109

HP-41C CLUB

ANYONE who owns an HP-41 should join John Lanphere's "Audio 41" - "Audio Engineering Software Library Club." The cost is $15.00. (See Tech Topic Volume 9, No. 4, for full details.) Membership in the club will return the investment many times over.

John sent us a list of the new programs added recently (added to an already valuable list of programs):

No. 41015 - Russ Berger's "MTP" (Mix Temporal Patterning)

Documentation. $7.50
10 Mag Cards $5.00

No. 41016 - Russ Berger's "NRM" (Normal Room Modes)
(Axial Resonant Modes of a Room)

Documentation. $7.50
6 Mag Cards. $3.00

No. 41017 - John Lanphere's "Locate"

A seating area aiming program. Locates 12 points down the center and around the edges of a seating area. Enter position and elevation information about the seating area--it gives back angular aim, range, elevation and dB_{gain} distance relative to center rear seat. Similar to the others but it is particularly friendly to the architect and the installer.

Documentation. $7.50
6 Mag Cards. $3.00

No. 41018 - Mark Laffin's "DISSPKR"

A complete distributed speaker layout program--enter room length, width, ceiling height, listener ear height--it gives back spacing information for square and hexagonal patterns--edge-to-edge, minimum overlap and center-to-center.

Documentation. $5.00
4 Mag Cards. $2.00

No. 41019 - Tom Bouliane's "VENT"

Program to design Thiele aligned Vented speaker boxes.

Documentation. $8.00
8 Mag Cards. $4.00

John's address is: Audio/Video Design Services
P. O. Box 6201
South Bend, IN 46660

John wanted us to mention that the "HP Users Club" has three useful audio programs available written by Syn-Aud-Con graduates:

No. 01033C - Rick Chinn's program to design Bridged Tee Pads and Equalizers. Price. $8.00

No. 01015C - Tom Bouliane's System Design program. Price. . . . $10.00

No. 01451C - Tom Bouliane's "Vented Box" program. Price $12.00

TIME

The second has been chosen as the fundamental unit of time in the metric (S.I.) system. It equals, by definition, the 1/86,400 part of a mean solar day.

It is of interest to note that in "astronomical time" the day commences at noon and is a 24 hour day. The solar year is defined as:

> The time in which the earth makes one revolution around the sun. Its average time, called the mean solar year, is 365 days, 5 hours, 48 minutes, and 48 seconds (nearly 365-1/4 days).

Time is the component in the metric (S.I.) system that successfully defies being decimalized.

THRESHOLD HEARING VS HEARING AT HIGHER LEVELS

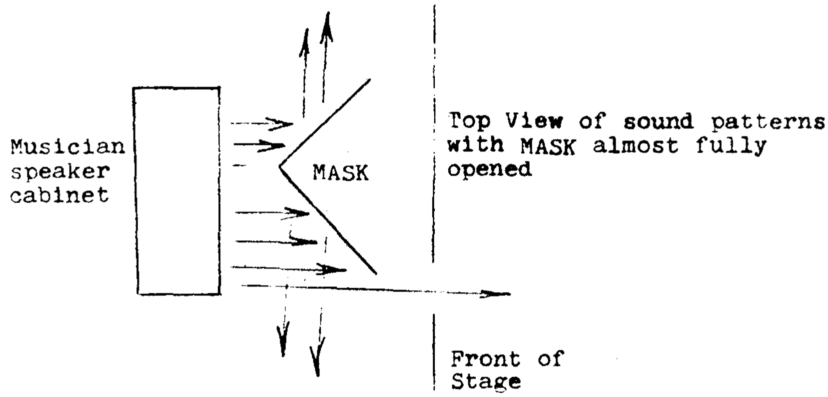
We recently had the privilege of having Mr. and Mrs. V. M. A. Peutz traveling with us in our motorhome from Minneapolis, Minnesota, to our farm in southern Indiana.

Among the myriad subjects discussed was the fact that persons with hearing losses at the *threshold* level may, indeed, exhibit nearly normal hearing at higher sound levels. As I understand it, this is due to the cochlea not using the same mechanism at all sound levels.

During the discussion, I recalled asking Wally Heider about the possibility of performing hearing checks on mixers at Wally Heider Studios. (We attended "A Day with Wally Heider" sponsored by the Los Angeles AES chapter.) Richard Heyser was present and said, "The threshold curve is not the same curve at higher levels." If anyone can suggest a reference in the literature to study this phenomenon further, please let us know.

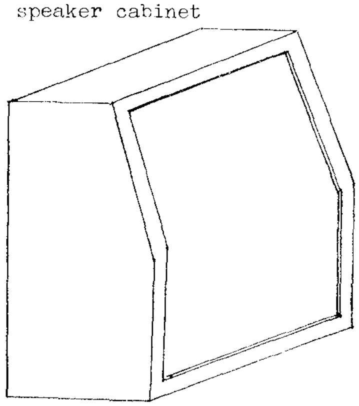
MASK – Musician's Amplification Sound Kontroller

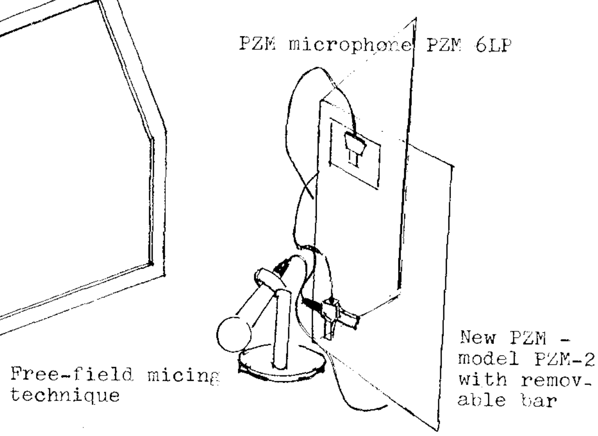
STEVE PAOLUCCI of RAM Electra-Acoustics sent an interesting use of the PZM® in a MASK - Musicians' Amplification Sound Kontroller. If you should want to contact Steve for more information, call 518/371-3965.



The Musicians' Amplification Sound Kontroller (MASK) is a device developed for the control of extreme SPLs off the stage by high-powered musicians. Oftentimes in live performances, musicians' stage volume exceeds the room sound system level causing an uncontrollable problem for the sound technician (mixer).

MASK shown in front of Musician speaker cabinet





Capsule of New PZM-2 should be mounted at the junction of the center of the plate horizontally and the floor

The purpose of the "MASK" is to deflect and control the direct sound from the front of the musicians' speaker system from shooting off the stage at excessive levels without interfering with band members' ability to hear stage sounds and without interfering with the musicians' needed levels for special effects such as "sustaining feedback" and overdriving.

As shown in the drawings, the "MASK" is constructed of two (2) identical pieces of 1/4" clear plexiglas connected on the long ends by a piano hinge. The hinge allows the "MASK" the ability to store and travel easily and also lets the sound technician control the amount of sound that is deflected by the "MASK" by opening and closing it in front of the musicians' speaker system.

I used 2-1/2 feet by 1-1/2 feet pieces of plexiglas in our "MASK" as they, when opened, would cover over the front of most speaker systems. Reflective tape about 1" wide was placed along the top and sides as to make the "MASK" more visible in low stage lighting. "MASKS" can also be stacked for tall systems or suspended on microphone stands. Microphone adapters are located on the front and rear for easy mic placement and to save on the number of mic stands used. Lately, the use of CROWN PZM®s has been explored with favorable results. The CROWN PZM-6LP seems to work best for high-powered guitars. The estimated cost of constructing a "MASK" is approximately $20.00. In a standard installation I recommend five (5) "MASKS."

DECIBEL REFERENCE LEVELS

Some reference levels in most common usage are:

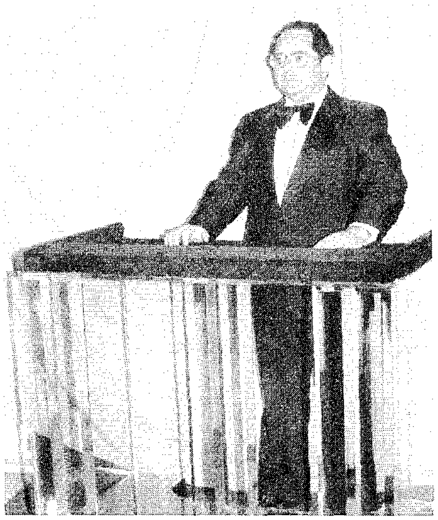
1. dBw — where the reference power is one watt
2. dBm — where the reference power is one milliwatt
3. dB*p* — where the reference power is one picowatt
4. dBV — where the reference voltage is one volt (taken to be an open circuit unless specifically stated otherwise)
5. L*p* — where the reference sound pressure is twenty micropascals

$$Lp = 20 \text{ LOG} \left(\frac{X \text{ pa}}{20 \text{ mpa}}\right)$$

6. Lw — where the reference power is dB*p*

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

7. A commonly used reference voltage is 0.775V. This springs from a misuse of the dBm concept and a great care must be exercised is using *levels* referenced to voltages to insure that it is known exactly what reference voltage is used and whether or not it is an open circuit voltage or associated with a resistance. Further difficulties with voltage references stem from the failure to specify if the values are peak, RMS, or average readings.



PZM® PODIUM

Pictured here is LARRY ESTRIN with his latest creation for the recent academy awards presentation, a "PZM®P" (a PZM® podium).

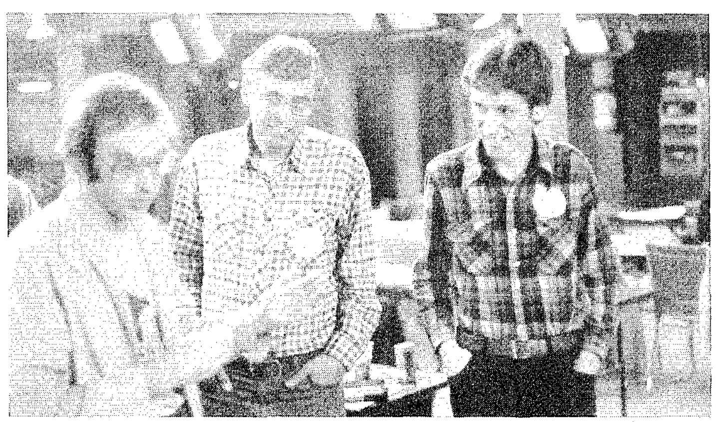
Those of you who viewed the program on television may have noticed that the host for the evening, Johnny Carson, specifically mentioned the "new microphones built into the podium" and the audience's enthusiastic acceptance of the sound being produced.

As we often say in class, we have had 50 years to learn how to use and misuse conventional microphones. We haven't had 50 months to learn how to use the Pressure Zone Microphone.

HP-41C - DEAD BATTERIES

BRUCE JACOBS of Fargo, North Dakota, writes us about the following interesting observation after finding that he often had to replace batteries in his HP-41C after traveling by air:

> Shortly thereafter, I acted on a hunch and checked the calculator right after it went through an airport X-ray. I found it on, and in a strange state where most of the keys were re-assigned (even the ← key which is not supposed to be re-assignable). I could only clear the machine by removing the battery.
>
> I now suspect that the airport X-ray devices can on occasion throw the HP-41C into a crazy state where it stays on, and possibly even draws excess power by continuously running in circles.



Bruce Jacobs, right, along with Herb Chaudiere, middle, consultant from Seattle, listening to Bob Hagenbach of Indianapolis during the February Loudspeaker Array Workshop.

SAN FRANCISCO CLASS

LEDE™ "DOWN UNDER"

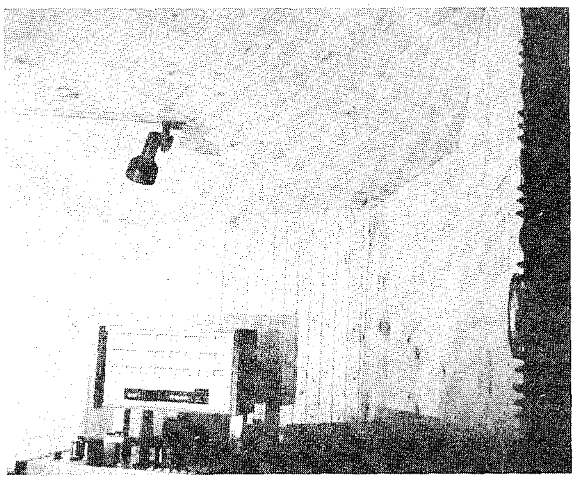
LEDE™ technology is now represented in Tasmania (get out your atlas and look "down under").

We are reproducing NICK ARMSTRONG's letter and the pictures of his control room as we found both items fascinating. We are impressed with their clear understanding of how to apply LEDE™ principles and their forward looking attitude to technology as a whole.

It would be our expectation that we'll be hearing a great deal more good news from such active, informed designers.

Dear Don: *26th February, 1982*

Thank you for your letter of 2nd February. Please find enclosed some professional and amateur black and white additional photographs which may be more explanatory of our L.E.D.E. (modified) control room. Also enclosed is a copy of the plan of the control room.



Some of the points of interest:-

1. *Our monitor speakers are Tannoy HPD (15" dual concentric). UREI one third octave room equalizers are used.*

2. *They are suspended by chains from the concrete slab ceiling with rubber isolation at the junction of the speaker cabinets.*

3. *As you will see from one of the photos, the front face of the console (facing speakers and glass doors) has Sonex on it. This was installed at the building stage so we haven't tried it without Sonex.*

4. *As yet, we haven't tried throwing Sonex over the console top (work face) during playback but we will try.*

5. *Would it be worth hanging Sonex over the glass doors during mix-down?*

6. *Calculated (not measured) initial time delay of the studio is 5.53 mS. The I.T.D. of the control room is 13.1 mS.*

7. *We have made provision (burlap covered open sections) for a bass trap in the ceiling rear. You may see this in one of the photos. The main problem in the control room is theoretically at 63.7 Hz due to the parallel concrete floor and ceiling.*

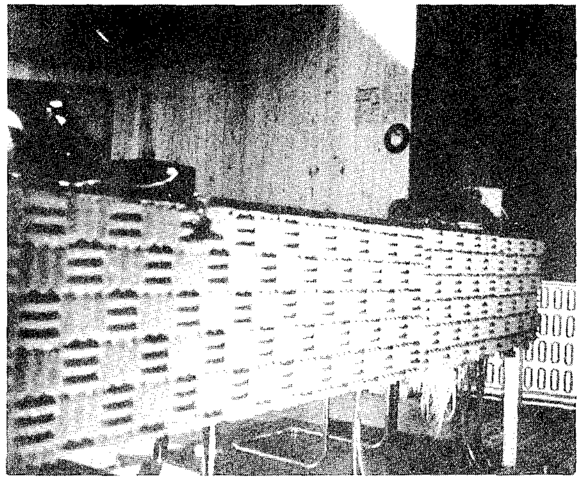
8. *Floor coverings:-   front half - carpet*
   rear half - vinyl sheet

9. *As yet we still haven't had a T.D.S. test although the room has been 1/3rd octave equalized with pink noise.*

We look forward to any comments or articles that may be forthcoming.

Incidentally, also enclosed is a thin section of Australian Sonex 4" foam manufactured locally by Illbruck. It is the type we have used exclusively in our control room. Because it is laminated with shorter "tongues"

Continued next page.....



Sonex placed on the back of the console.

(profile) we believe it may not be as good as the American equivalent, particularly in the low frequency end. Would you be interested in doing some test measurements? If so, what is the smallest area (size) we can forward to you?

All the best, NICK ARMSTRONG, SPECTANGLE PRODUCTIONS, 40 George Street, North Hobart, Tasmania 7002

P.S.: We have just purchased an Apple II Plus (48K) micro computer. (5" discs). We have recently tried the Basic programmes for Room Modes and Hermholtz Resonators and RT_{60} *as presented in October and November 1981 "dB" magazines. If you know of any source of similar programmes (maybe like your new HP41 cards), we would be very interested. (The computer was originally purchased for debtors and book keeping!)*

A BREAKTHROUGH IN DIGITAL ADVERTISING

A new "digital" equalizer is being "hyped" in the audio press as vastly superior because it can generate extremely narrow rejections at very rapid rates. I would simply remind those of you still not swept off your mental feet by this advertising feat that:

1. The "ringing" of any filter, be it digital or analogue, is determined *solely* by its *bandwidth*.

2. Instantaneous adjustment of any device to the total sound field will always adjust the direct sound field's response even when the problem lies in a *time delayed reflection*, with all the attendant unnaturalness that implies.

In essence, what is claimed for this "digital equalizer" is that it can *incorrectly readjust* the system's transfer function *faster*.

Reviewer Fieldmouse: "I think the digital filter is mar-vel-lous-s-s-s."

Engineer Breakwind: "Of course, I said so!"

AUDIO PORNOGRAPHY

Pornography is defined by Webster as: "Description or portrayal of prostitutes or prostitution." The truth could hardly be more "prostituted" than it is in current popular audio magazines by their "resident" experts and authorities.

A recent "bubbling" review of a "digital" replacement for equalizers leaves me with the feeling that these reviewers, I fear, are never going to get their act together. It's, alas, audio pornography.

At left is a picture of a satisfied listener to a "digital" equalizer.

SPECIAL COMMUNICATIONS SYSTEMS

For those persons with a substantial hearing loss, higher than normal acoustic gain and lower than normal articulation losses of consonants are a necessity. A most practical way to satisfy both requirements simultaneously is to resort to personal receivers issued to each individual having the hearing difficulty. We will be reporting on a number of these systems in future Newsletters. To begin our discussion of these *communication* systems, we offer RUSS O'TOOLE's excellent summary of a recent job he undertook to provide a simultaneous translation system for a church which utilized the *typical* "hard of hearing" techniques and equipment.

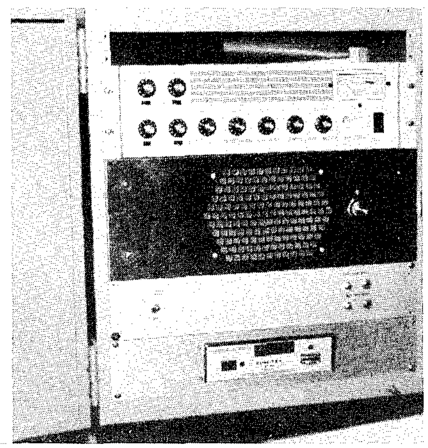
Korean Presbyterian Church

The Korean Presbyterian Church has a very unique communications problem. Someone was kind enough to suggest the church call a Professional Sound Contractor who specializes in church installations. We eagerly responded to the Pastor's invitation to meet with his committee.

The Church has a large auditorium (sanctuary) capable of seating around 7-800 persons including a large balcony. Typical attendance averages 6-700 per Sunday service.

All of the services are conducted in the Korean language; however, about 20% of the congregation is comprised of small children. It was the desire of the parents for the children to hear the services not in Korean but in the English language. Hence, the need for a "simultaneous interpretation system!".

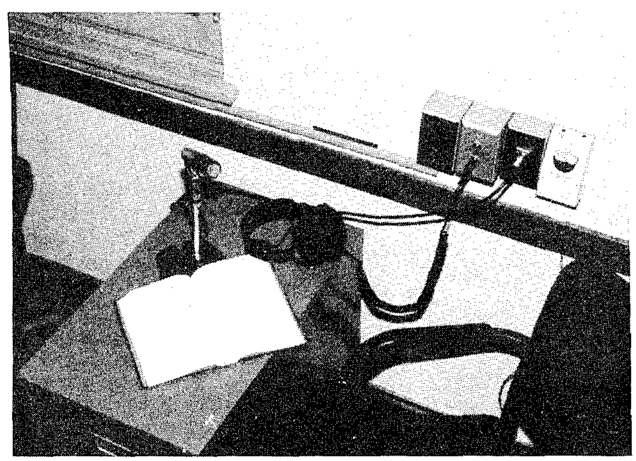
Korean Presbyterian Church, Cicero, Illinois.



Equipment cabinet with wireless transmitter.

Due to the large seating area, 60' x 100', and the age of the building, we felt the often used concept of a wired loop system to be not practical.

Vertically polarized "J fed" antenna.



Left to right: Headphone volume control, headphone jacks, mic input receptical, remote metering showing power on RF, and indicator of modulation.

Continued next page.....

One manufacturer of an AM wireless hard-of-hearing system was considered but limited range from the transmitting antenna plus the tendency for atmospheric and induced electrical interference caused us to reject that solution.

We have been using a very effective and low cost FM wireless microphone system for our church customers and were aware of a small portable crystal controlled FM receiver being manufactured for auditory training purposes.

Although the FM transmitter puts out only 100 mw, fed into a simple "J fed" antenna system we have been able to easily cover distances of over 300'.

A demonstration of this system plus the clarity and range of reception convinced the church that this was the best way to go.

To help keep the cost of the system as low as possible, we built up a series of three-legged patch cords allowing them to use up to three stethoscope headsets from the same receiver.

Another requirement of the system was that the earpieces were to block out as much ambient sound as possible. A manufacturer of soft pliable ear pieces provided us with a very excellent set of headphones at a modest cost.

The church has a closed circuit television system and provided a video monitor in the translation room where the translator listens to the services from the PA system (in Korean) and then rebroadcasts the services to the rack mounted FM transmitter.

The system has been in service for nearly four months now and the church's people are well satisfied with their investment.

When planning any type of a wireless system, it is well to select frequencies which are known to be clear of any possible interference. In many cases we have successfully used the Audio carrier frequency of unused VHF TV channels. Where you have a channel 5 and a channel 7 in the same community, channel 6 is usually a pretty safe bet!

FLYING ENGINEERS

BILL KESSLER, consultant with Kessler and Gehman Associates, Ltd., in Gainesville, Florida, is a Flying Engineer. We know many Syn-Aud-Con graduates are--MANNY MOHAGERI, JIM CAREY, AL SEIPMAN and BOB LIN to name the first ones that come to mind.

Bill wants all Flying Engineers to know about Flying Engineers International.

Flying Engineers International was founded in 1964. The primary objectives of the organization are to promote flight safety and the utility of light aircraft by professional engineers. An additional goal is to act as a recognized political action group to express views before the Federal Aviation Administration, the Federal Communications Commission and Congress on those issues of importance to general aviation which is representative of flying professional engineers.

One-day regional meetings are held approximately 4 times a year and a three-day annual meeting is held in May or June. During the annual meeting new officers are elected, technical sessions are held and the meeting is concluded with a banquet on the evening of the third day. Newsletters on timely subjects are mailed to the membership during each quarter. Additionally, an annual issue of HANGER FLYING is published and mailed to the membership.

A recent annual meeting of the Flying Engineers for 1982 was held in Chattanooga, Tennessee, at the Best Western Airport Inn during June 11, 12 and 13. The place and time of future regional meetings will be announced.

The Flying Engineers International is a professional organization. Therefore, dues and other expenses incurred attending technical meetings are deductible. Annual membership dues are a modest $15.00. For additional information interested persons may contact: W. J. Kessler, President-Elect, Kessler and Gehman Associates, 1511 N.W. Sixth Street, Gainesville, FL 32601 or phone: 904/376-3157.

MINNEAPOLIS CLASS

CALCULATING PERCENTAGES AND RATIOS

dB for % Below Reference

$$dB = M \, LOG_b \left(1 - \frac{\%\ change}{100}\right)$$

dB for % Above Reference

$$dB = M \, LOG_b \left(\frac{\%\ change}{100} + 1\right)$$

Examples

(1) A harmonic is 1%[*i.e.*, 99% below reference]

$$dB = 20^* \, LOG_{10} \left(1 - \frac{99\%}{100}\right) = 40\ dB$$

**Voltage like ratios*

(2) An acoustic signal is reflected off of a surface that is 80% absorptive. How many dB does the reflected signal drop?

$$dB = 10^{**} \, LOG_{10} \left(1 - \frac{80\%}{100}\right) = 6.99\ dB$$

*** Power like ratios*

(3) We raise the input voltage to a loudspeaker by 30%. How many dB should we add to its SPL?

$$dB = 20 \, LOG_{10} \left(\frac{30\%}{100} + 1\right) = +\ 2.28\ dB$$

FILTER BANDWIDTH DESIGNATIONS

Given:

A filter *labeled* 1/3 octave. Find its bandwidth as a percentage of its center frequency (f_C).

Solution:

Make f_C = 100 Hz. This allows the bandwidth in Hz to equal the percentage bandwidth of f_C. Remember, filters *labeled* 1/3 octave are in actuality 1/10 decade spacings of f_C and, further, that the labels are rounded values of the true f_C which is a 10 series set of Renard numbers. (See *SOUND SYSTEM ENGINEERING*, page 174, Table 9-4)

With these qualifications in mind, we can use the general case equation for logarithmic spacing intervals.

$$f_D \text{ at } ND = L_f\left(\frac{H_f}{L_f}\right)^{\left(\frac{1}{N_T}\right)^{ND}}$$

Where: f_D at ND is the desired frequency at the desired interval

N_T is the total number of intervals on the scale chosen

L_f is the lowest frequency on the interval scale (H_f is an ND = 0)

H_f is the highest frequency on the interval scale

Take 100 Hz as our f_C and take 80 Hz as L_f and 125 Hz as H_f--the next lowest and next highest intervals on the 1/10 decade scale (their exact Renard numbers to two places are 79.43 Hz and 125.89 Hz). We then create 4 intervals between these frequencies--1/6 octave intervals because 1/6 octave lower than 100 Hz and 1/6 octave higher than 100 Hz creates a bandpass 1/6 + 1/6 = 2/6 = 1/3 octave wide centered on 100 Hz. Thus, N_T = 4, the interval below 100 Hz = 1.0, and the interval above 100 Hz = 3.0 (100 Hz = 2.0).

$$f_D \text{ at } ND_1 = 79.43\left(\left(\frac{125.89}{79.43}\right)^{1/4}\right)^1 = 89.12\ Hz$$

and

$$f_D \text{ at } ND_3 = 79.43\left(\left(\frac{125.89}{79.43}\right)^{1/4}\right)^3 = 112.20\ Hz$$

(112.20 Hz - 89.12 Hz) = 23.08 Hz & % Bandwidth

FINDING THE "RENARD SERIES" FOR FRACTIONAL OCTAVE SPACING

Renard numbers are equally spaced intervals on a logarithmic scale.

In the Renard number system, one octave is specified as the 3-1/3 series (i.e., $10^{\left(\frac{1}{3^1/3}\right)}$ is the multiplier, m, used to increment each interval).

To find any other Renard series, multiply 3-1/3 by the reciprocal of the fractional octave desired. (i.e., 1/6 octave = 6 X 3-1/3 = 20 series; 1/10 octave - 10 X 3-1/3 = 33-1/3 series.)

General Case Equation

$$f \text{ intervals} = (L_f \cdot m)\, m \ldots\ldots m_n$$

<u>EXAMPLE</u>

To obtain 1/3 octave intervals from 100 Hz up:

$$f \text{ intervals} = (100(10^{1/10})) \times (10^{1/10}) \times (10^{1/10}) \ldots\ldots (10^{1/10}{}_n)$$

See *SOUND SYSTEM ENGINEERING*, pages 173 and 174, for the listings of preferred frequency *labels* and preferred frequency exact values. (Tables 9-3 and 9-4)

CONVERTING ACOUSTIC REFERENCE LEVELS

1. <u>OLD STANDARD</u>

   10^{-13}w (10 LOG 1w) - (10 LOG 10^{-13}w) = 130 dB

   (i.e.: The reference is 130 dB *below* one watt)

   <u>NEW STANDARD</u>

   10^{-12}w (10 LOG 1w) - (10 LOG 10^{-12}w) = 120 dB

   (i.e.: The reference is 120 dB *below* one watt)

2. <u>OLD STANDARD</u>

   dB - SPL ≈ dB - PWL at 0.282 ft (0.08595 m)

   <u>NEW STANDARD</u>

   dB - SPL ≈ dB - PWL at 0.282 m (0.925ft)

3. To convert old Standard power level to new Standard power level, subtract 10 dB (i.e.: move it 10 dB closer to one watt)

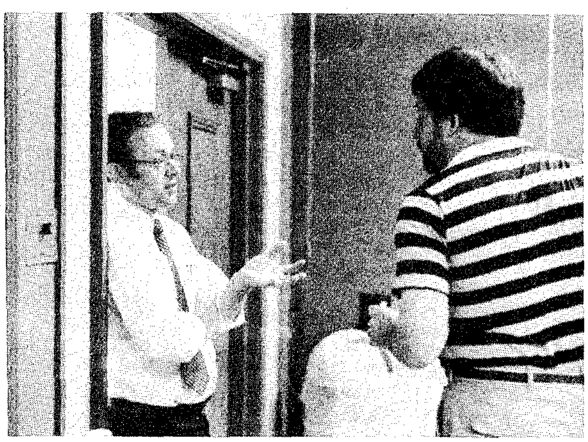
4. To convert new Standard power level to old Standard power level, add 10 dB (i.e.: move it 10 dB further from one watt)

5. Old Standard dB - SPL at new Standard reference distance (0.282m) assuming Q = 1 and 1 watt.

   $$130 + 20 \text{ LOG} \left(\frac{0.08595\text{m}}{0.282\text{m}}\right) = 119.68 \text{ dB}$$

6. New Standard dB - SPL at old Standard reference distance (0.08595m) assuming Q = 1 and 1 watt.

   $$120 + 20 \text{ LOG} \left(\frac{0.282\text{m}}{0.08595\text{m}}\right) = 130.32 \text{ dB}$$



Warren Ediger (back to us) talking to Ed Lethert

SMILE

An architect is said to be a man who knows a very little about a great deal and keeps knowing less and less about more and more until he knows practically nothing about everything; whereas, on the other hand, an engineer is a man who knows a great deal about very little and who goes along knowing more and more about less and less until finally he knows practically everything about nothing. A contractor starts out knowing practically everything about everything but ends up by knowing nothing about anything due to his association with architects and engineers.

Contributed by WARREN EDIGER
Omaha, Nebraska

WHEN TO VARY L_W VS Q TO CONTROL L_D

One of the more interesting questions to consider in the design of a Loudspeaker Array, LSA, is the adjustment of the desired direct sound level, L_D, without detrimentally affecting the ratio L_D/L_R. (Where L_D is the direct sound level and L_R is the reverberant sound level.)

The two limiting cases can be handled the easiest. In the case where $\Delta dB = 0$ it is essential to vary Q in order to adjust L_D with the least disturbance to L_R. (Where ΔdB is the number of decibels L_T is below the Hopkins-Stryker calculation at $2D_c$.) In the case of $\Delta dB = 6.0$ or greater, varying L_W causes no disturbance as there is no reverberant sound field. (Where L_W is the sound power level in dB for the device providing L_D at D_x (ref. 10^{-12} watt).)

If we assume a $\%AL_{CONS}$ of 15 as our design goal, a change in L_D/L_R of 1 dB raises our $\%AL_{CONS}$ to 18.88. This would suggest a ΔdB of at least 5.0 if we are to safely manipulate L_W to obtain our desired L_D without unduly influencing L_R. In recognition of how grey an area exists here, perhaps a $\Delta dB = 3$ would be a good area to begin to exercise caution regarding this practice. Syn-Aud-Con is particularly receptive to any data from the field where varying power levels instead of Qs either worked noticeably well or obviously poorly.

It must always be kept in mind that the underlying cause of the ΔdB effect is the absence of a steady reverberant sound field. The L_T is being raised above the L_D by a transient sound field (after a small number of discrete high level reflections) and not a steady, mixing, homogenous one. V.M.A. Peutz has utilized the IRCAM variable acoustic (0.5 to 5.0 secs) concert hall to investigate the detailed role of V, h, RT_{60}, absorption and diffusion on the measured ΔdB and we will be publishing some of this work in the near future.

HAROLD W. LINDSAY - AN AUDIO GREAT

Harold Lindsay with John Mullin

HAROLD W. LINDSAY of Ampex fame passed on in April this year. Syn-Aud-Con made no secret of our immense respect for and admiration of Harold Lindsay. A totally creative man with that enviable talent of conceiving the as yet totally unknown and then applying hard-headed scientific engineering principles to the dream's fulfillment, he inspired a generation of talent to follow his lead. Enthusiasm, gentleness, integrity and artistry are natural descriptions of the man.

As we have noted in the past, Harold lived an exceptionally successful life in every meaning of the word "successful." The word "pioneer" is often used to describe those present at the beginning of an event. Harold literally generated the tape recorder industry in the United States, including video tape recording. Syn-Aud-Con considers it a rare privilege to have been included among his friends and live more meaningful lives because of it.

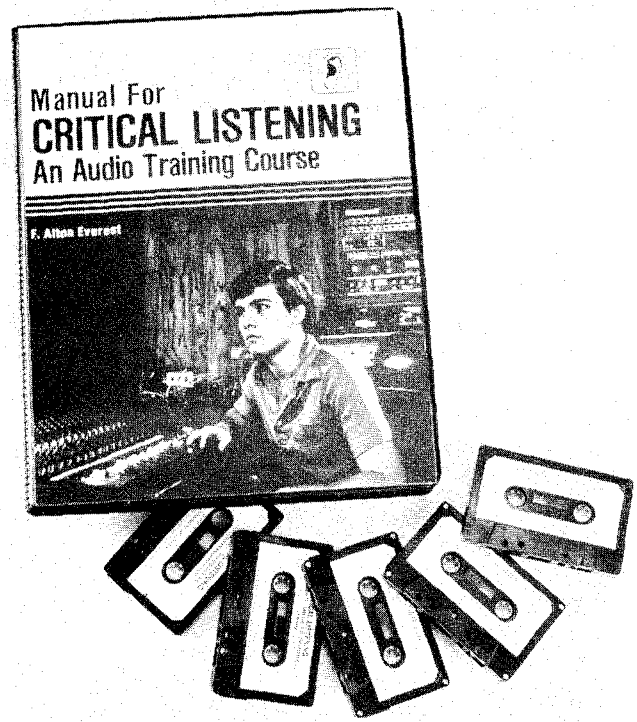
"MANUAL FOR CRITICAL LISTENING"

An Audio Training Course

We have just received our copy of F. Alton Everest's *Manual for Critical Listening = An Audio Training Course.* (Alton Everest is an internationally known author, lecturer, engineer, and acoustical consultant.)

This 106-page manual accompanied by five audio cassettes is a truly outstanding accomplishment. It is designed for anyone whose career will depend upon the abilities of hearing perception. Through his unique *Talk/Tech* method, Mr. Everest introduces the student to the most common sound characteristics and irregularities, as well as basic sound-shaping techniques. The student can simultaneously hear the narrator's *talk* and view the *technical* aspects of the topics through drawings, pictures, diagrams, and easy-to-read explanations of the recorded examples.

Every lesson is concluded with an aural test to help measure your ear training process. And because all the examples are on cassette tapes, any lesson can be repeated as many times as necessary to insure complete understanding before proceeding to the next one. The textbook and cassettes pull together, into one convenient package, many simulated audio situations that would normally take months or even years to encounter and define in a normal day-to-day working environment.



Lessons include:

1. Estimating the frequency of sound
2. Estimation of sound level changes
3. Estimating frequency band limitations
4. Frequency response irregularities
5. Judgment of sound quality
6. Detecting distortion
7. Reverberation effects
8. Signal vs noise
9. Voice colorations
10. Listening with discernment
11. Supplement

The retail price for this manual complete with cassettes is only $129.95. Quantities of from two to nine have a discount of 20% off and in quantities of ten or more, a 40% discount applies.

For those of you wishing to take advantage of the longer discount on this exceptional teaching tool, Syn-Aud-Con will until September 1 accept your payment for the amount of $78 + $10 handling charge + Calif sales tax if a resident of California.

This represents approximately a 30% discount on the purchase of a single set. Those wishing to buy in quantities should contact the publisher, SIE Publishing, P O Box 4139, Thousand Oaks, CA 91359.

BOOKS OF INTEREST

The Science of Sound"

SCIENCE OF SOUND by Thomas D. Rossing, published by Addison-Wesley, is dedicated "to all those who have taught me so that I could teach others." This 637 page, 33 chapter book is well written and illustrated. I have never met Professor Rossing but the personality projected by the book comes across as a pleasant inquiring mind who loves audio and acoustics.

The key value I see in the book and the virtue that makes it a worthwhile addition to the library of any Syn-Aud-Con graduate is that it briefly introduces a massive number of technical concepts, identifies their source in the literature and thereby allows the reader to use this volume as an initial survey source.

A very useful balance between acoustics and music is achieved and, while "science" is in the title, this volume expresses a great deal of empathy for the "art" of sound. Maximum use has been made of illustrations and the short incisive chapters allow, yea, encourage browsing in the book during leisure time.

Don't look here for derivations or extensive explanations. This book skims the surface but has the virtue of skimming over relevant frequently overlooked topics. Once the topic is identified there is nothing to stop the reader from pursuing it further in more detailed literature. Professor Rossing is, in the best sense of the word, a generalist and has provided a most useful tool in the publication of his book.

CLASSIFIED

STOLEN EQUIPMENT: The following list of equipment was stolen this spring from Dave Andrews of New York City:

1 - GenRad 2512 FFT - Serial No. 302
1 - Hewlett Packard 3325 Synthesizer - Serial No. 1748A01536
1 - Hewlett Packard 3580 Spectrum Analyzer - Serial No. 1312A00140
1 - BDT2 - Serial No. 11197
1 - Crown RTA^2 - Serial No. 21021

Should you have any information regarding the above equipment, please contact Dave Andrews, Andrews Audio Consultants, 347 West 39th Street, New York, NY 10018. Ph: 212/674-6934.

* * * * * * * * * * * * * * *

FOR SALE: Books - *HOW TO BUILD A SMALL BUDGET RECORDING STUDIO FROM SCRATCH* by F. Alton Everest. TAB 1166. Price. $10.00

THE MASTER HANDBOOK OF ACOUSTICS by F. Alton Everest. TAB 1296. Price. $15.00

CONCEPTS OF ARCHITECTURAL ACOUSTICS by David Egan. Price $41.00

The above prices include postage paid if shipped within the continental United States. If you are interested in obtaining any of the above, please contact John Lanphere, Audio/Video Design Services, P. O. Box 6201, South Bend, IN 46660.

* * * * * * * * * * * * * * *

FOR SALE: Altec 9014 1/3 octave passive filter set. Price $600.00

Contact Tim Tommerson, DBT Sound, 3018 South 17th Street, Moorhead, MN 56560.

* * * * * * * * * * * * * * *

FOR SALE:

1 only HP-8050A Real Time Analyzer. $1450.00
4 each HP-8062A Impulse Sound Level Meter . . . .$ 225.00 each
3 each HP-15109B Calibrated Microphones . . . .$ 200.00 each
1 only Communications Company RT_{60} Reverberation Time Analyzer. .$ 250.00
2 each Altec 1680 Digital Pink Noise Generators - NEW . . . . . .$ 100.00 each
2 each IVIE IE-10A Octave Analyzer and Sound Level Meter with Charger. . . . .$ 590.00 each
1 only IVIE IE-30A Spectrum Analyzer with case and 1133 Microphone, Charger and Manual - NEW $3000.00

Contact Lee Ritterbush at 303/471-8430.

* * * * * * * * * * * * * * *

FOR SALE: Hewlett Packard personal computer CRT, printer, tape drive, keyboard, memory in one complete portable unit. Complete with owner's manual and basic training guide. Nearly new - only 50 hours training time. Call 616/243-2094 or 245-1938 after 5:00PM.

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WANTED: TDS equipment. Contact Charles Bilello, 258 Fairlawn Avenue, West Hempstead, NY 11552. Ph: 516/489-7463.

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POSITION WANTED: Acoustic Consultant, Project Manager, or Audio Systems Engineer. I am a widely experienced technical and construction project manager, specialized in electro-acoustic and audio/video systems design. I have specified and authorized up to $5.5 million in construction materials on past and current projects and have worked with architects, consulting engineers, contractors, and owners with excellent rapport. I have designed and installed "state of the art" technical systems; demonstrated effective analytical, communicative and managerial skills; trained technical and administrative staffs; interfaced in an extremely successful manner with customers; and have been a guest lecturer in acoustics to Vanderbilt University. I am one of the original 10 licensees for TEF-TDS acoustic measurements. Contact: Richard R. Lee, 932 Colonial Drive, Baton Rouge, LA 70806. Phone: Area 504/ 925-0491.

SYN-AUD-CON SPONSORS

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

Personnel from these manufacturers receive Syn-Aud-Con training which provides still another link in the communications circuit between the ultimate user and the designer-manufacturer of audio equipment. They are "in-tune" with what a Syn-Aud-Con graduate needs.

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