SYNERGETIC AUDIO CONCEPTS

newsletter

VOLUME 6, NUMBER 3
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SYNERGETIC
Working together; co-operating, co-operative

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

EXCHANGE OF IDEAS

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

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

TABLE OF CONTENTS

PAGE

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SYNERGETIC AUDIO CONCEPTS

TDS-ETC

The time delay spectrometry patent will be ten years old this coming September. Dick Heyser's remarkable paper on TDS was published in the *Journal of the Audio Engineering Society* in October, 1967. (The patent application was filed in January 1968.)

In May of 1969 I first demonstrated in an AES Technical Session the use of 1/3-octave real time analyzer (made by Hewlett Packard) as a way to observe the total sound spectrum as I equalized a sound system. The past ten years has seen the promise of 1/3-octave real time analyzers fully developed so that today the engineer needing such a device has literally a dozen choices ranging in price from just under $1,000 to over $20,000.

The decade of the 1980s will become, in our opinion, the era of TDS. Each new step further into the basic concepts of TDS reveals ever more clearly that the audio world is prepared to utilize this advanced tool. It has been observed that steam power was around for over a century without the development of a personal steam powered motor car. Investigation suggests that mankind as a whole was satisfied with the horse for short trips and the steam powered train for longer journeys *until* the advent of the bicycle near the end of the last century. The bicycle fad captivated the expanding city population and left them with a yen for rapid long distance capability in a personal transportation form. Bicycles also led to vastly improved roads and a willingness on the part of travelers to pay for roads. When all these prerequisites were in place the development of the modern motor car became logical and inevitable.

Similarly, the widespread use of 1/3-octave RTAs has resulted in an appreciation that while a meter can tell you the "level", an analzyer can show you that level by frequency, thereby leading to a new understanding of what is meant by the word "level". The 1/3-octave RTA is our most accurate, economical way to view the *total* sound spectrum.

The Total Sound Spectrum

When we view the "frequency response" of a sound system on the screen of a 1/3-octave RTA, we are seeing the total sound field (L_T). This sound field is the combination of the ambient noise level (L_N), the direct sound level (L_D), and the reverberant sound level (L_R). When there are less than 50 reflections per theoretical ray path for a signal decaying 60 dB after termination of the input signal, L_R may stand for the reflected sound level.

$$L_T = 10 \log \{\exp L_D/10 + \exp L_R/10 + \exp L_N/10\}$$

Where exp is the exponent of the base 10 such that $\exp L_D/10 = 10^{\left(\frac{L_D}{10}\right)}$

Which of these sound fields predominates can be adjusted to a significant degree by juxtaposition to the sound source.

The ambient noise level can be acquired by turning off all desired sources and measuring the ambient noise level by itself. Thus, if an accurate direct sound field can be obtained, the reverberant sound field can be calculated with accuracy. The direct sound field is easiest acquired by going, on axis, to the source at least 10 dB above the reverberant sound and then extrapolate by inverse square law to any required distance. Fine! I can do all of this with a 1/3-octave RTA.

The Role of TDS

TDS allows rapid acquisition of the direct sound field as a spectrum -- the direct sound field at every frequency from 200 Hz up. At low frequencies the law of physics still restricts us. Even more significantly, TDS allows us easy and rapid access to the spectrum of *selected* reflections. Those of you that witnessed early TDS demonstrations know that while such spectra were interesting for instructional purposes, they could also prove very confusing to the uninitiated.

TDS-ETC

Early this Spring Dick Heyser came down to our office in Tustin to show us something he called "Energy Time Curves", ETC. The expression on my face says it all, "Is this for real?". The calm collected expression on Dick's face followed his "Yup, you'd better believe it".

What Are We Looking At?

I am staring at the display on an FFT analyzer (FFT means Fast Fourier Transform" and is an analyzer that can transform mathematically from the time domain to the frequency domain given any impulse). Dick had our regular TDS sweeping on a loudspeaker in the room (one of our small Visoniks) and on the screen of the FFT was being displayed the *energy density* of each reflected spectrum (plus direct sound spectrum) vertically all separated *in time* on the horizontal axis. See Figures #1, #2, and #3.

In other words, we were viewing on the FFT-ETC the "end view" of all the spectra in the total sound field - L_N, L_{R_1}, L_{R_2} L_{R_n} etc. We were also seeing time misalignment, time warp, etc.

Then, Dick held his hand *over* the microphone. All ceiling reflection *spectra* dropped in level on the screen of the analyzer while all other reflected spectra -- walls, floor etc., remained the same. If he held his hand in back of the microphone, all rear wall reflected spectra dropped

TDS-ETC, CONTINUED

in level on the analyzer's display and the direct sound level rose, due to the hand reflection, while all other reflected spectra remained constant.

FIGURE # 1. TDS FREQUENCY DOMAIN REPRESENTATION

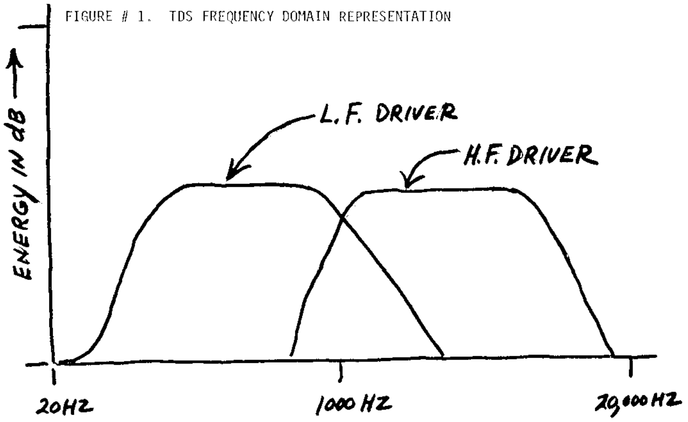


FIGURE # 2. ETC TIME DOMAIN REPRESENTATION

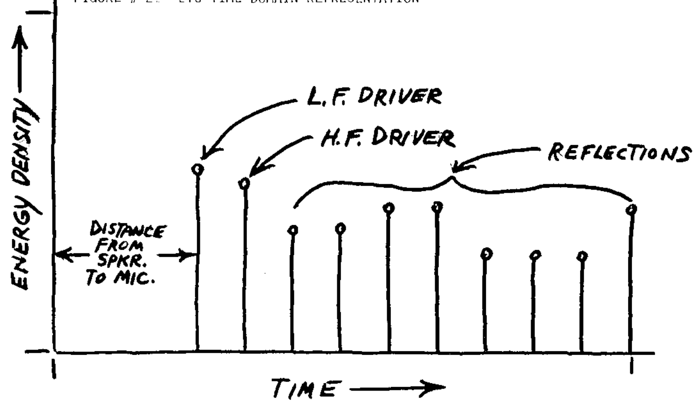
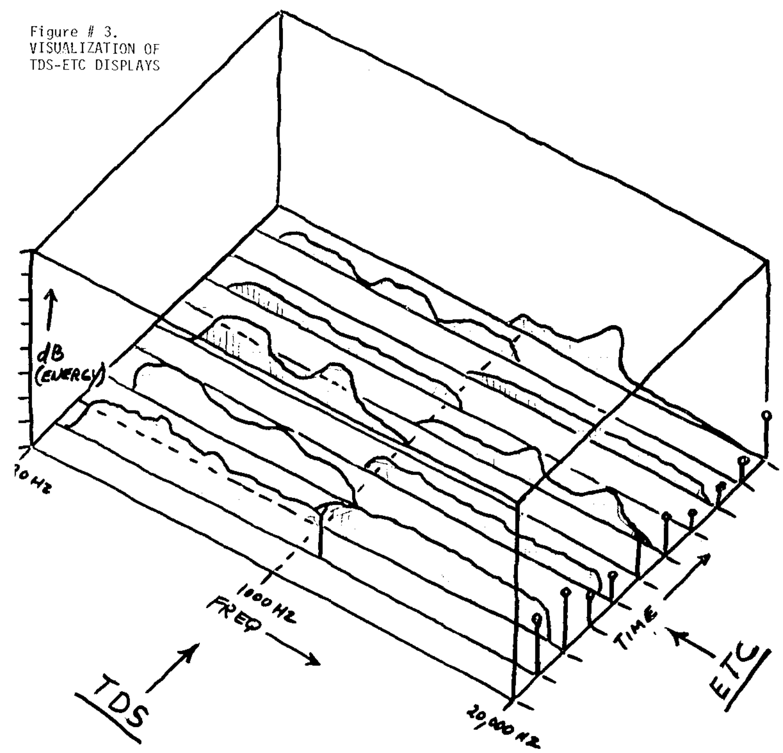


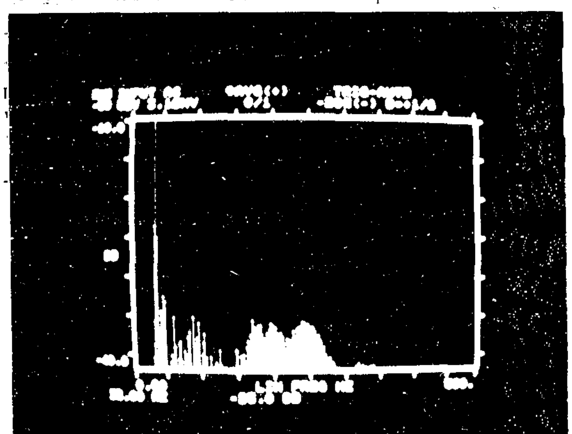
Figure # 3.
VISUALIZATION OF
TDS-ETC DISPLAYS



Think for just a moment what we are saying!

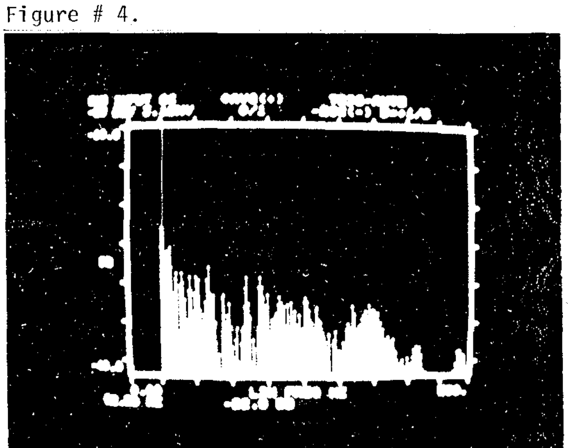
With a single TDS sweep we can observe *all* the spectra in the room for, as an example, the first 50 msec.

Figures 4 and 5 illustrate first a space with a high direct-to-reflected ratio and then a space with a low direct-to-reflected ratio. *Each* vertical line is the "end view" of a total spectrum.



HIGH DIRECT TO REFLECTED RATIO

Figure # 4.



LOW DIRECT TO REFLECTED RATIO

Figure # 5.

The blank space on the left side of each photo is the time delay between the source and the measuring microphone.

In each photo the lighted cursor (the vertical line the full length of the vertical scale) is set to the direct sound spectrum. The FFT frequency scale has now been transformed into a time scale and in these particular measurements, 1 Hz on the frequency scale equals .1 msec of time.

In these examples, .113 times the display frequency equals the distance in feet that the signal has traveled from the source. By obtaining the direct sound spectrum's travel time first and then subtracting it from any subsequent measurement, one can obtain the arrival time of each reflected spectrum relative to the direct sound.

As if this weren't mind-boggling enough, the TDS can then be placed into a mode that allows the TDS offset to be tuned to *wherever* the FFT cursor is set. Then on the screen of the TDS analyzer the spectrum of whichever of the 400 lines on the FFT is chosen by the cursor now appears.

TDS-ETC, CONTINUED

Let's summarize a little of what's being said here. We can now measure in essentially real time:

1. The arrival time to an accuracy in microseconds, if desired, of any sound, direct or reflected.
2. The *direction* from which it came
3. Its total energy density
4. Its spectrum (energy density vs frequency)
5. Any *velocity* changes in the transmitting media
6. Any time misalignments or warpage. See Figures #6 and #7.

Figure # 6. INTERACTIONS BETWEEN FREQUENCY TIME AND AMPLITUDE

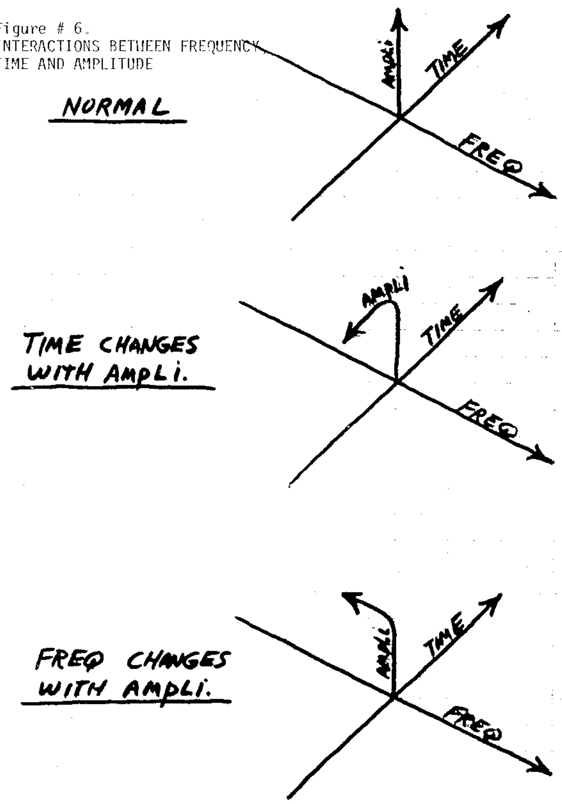
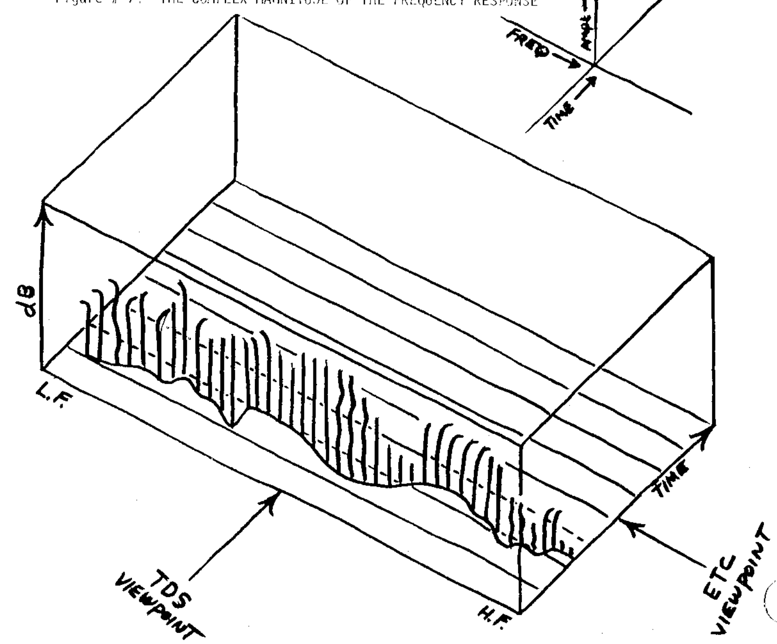
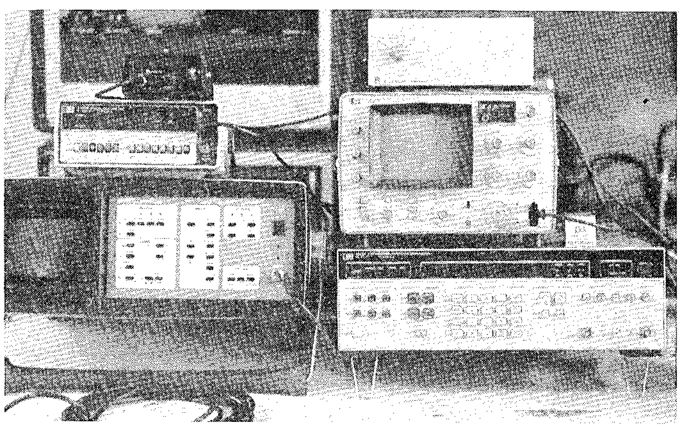


Figure # 7. THE COMPLEX MAGNITUDE OF THE FREQUENCY RESPONSE



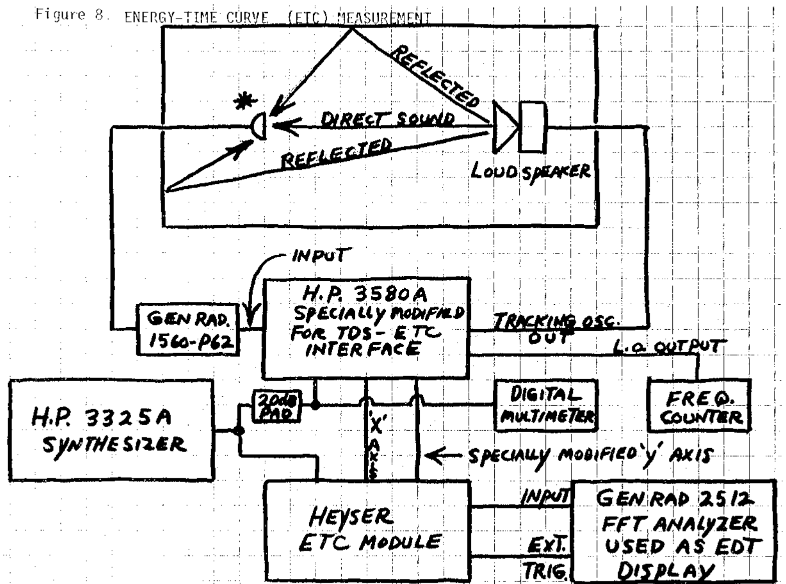


The photograph shows the equipment currently being demonstrated in the last 4 Syn-Aud-Con classes. Figure # 8 lists the equipment and diagrams its interconnection.

The Heyser ETC module is proprietary and only licensed TDS practitioners will be considered for the additional ETC license. We hope to announce by AES the availability of a license for ETC.

Needless to say, current Syn-Aud-Con classes are witnessing data never before available and all of us feel like we're riding a rocket sled on the technological test rail.

Figure 8. ENERGY-TIME CURVE (ETC) MEASUREMENT



*GENRAD 1560-P42 PREAMPLIFIER WITH B&K ½" TYPE 4136 PRESSURE RESPONSE MICROPHONE

TDS-ETC, CONTINUED

Lee Irvine, Acoustical Consultants Inc. in Salt Lake City brought to my attention a computer program used by the Norwegian Institute of Technology in Trondheim, Norway wherein they compute from drawings what we can now measure.

In fact, we can now measure with much more detail than is possible economically with the computer. The data, from one of Throndheim's brochure is instructive. See material reproduced here from their brochure.

What all of this means is that electroacoustics and architectural acoustics are leaving the world of two dimensions for a glimpse at the world of three or more dimensions.

It means that ten minutes after the apparatus is set up you are in a position to examine if you should get rid of a feedback in a sound system by use of an equalizer, absorption on a surface, or re-direction of that particular reflection because you can *see* where the energy causing the feedback *came from*. You can *see* what happens if you apply absorption or if you change the angle of that surface or if you apply a diffuser. You can *see* which reflection is interacting near regeneration and "tune" to it and observe if its anomolies are "on" feedback frequencies.

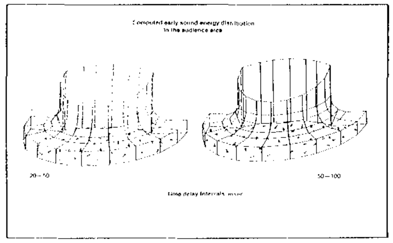
In studios and control rooms you can now *see* if you really have diffusion, if M_a is completely out of hand, if your monitors have time warpage, how sound transmission is occurring between spaces, etc.

<u>100 TDS Licensees</u>

Just a year ago there was 1 TDS licensee. Today there are today 100 worldwide. Will it be long before those without TDS-ETC in addition to their RTAs might as well shut off their hearing aids too because they're not going to be competitive in *professional* audio?

Most exciting of all, is the realization that meaningful, useful, relevant criteria can now be developed from the measurements being taken. I'd like to remind you that only Syn-Aud-Con graduates know about all this. The only two ETC analyzers in existence are Dick Heyser's and Syn-Aud-Con's. You have a unique piece of inside information. It's worth what your imagination and creativity can rise to. Knowing all of you and your potential we're excited.

A new tool for the prediction of room acoustics has been developed by Akustisk Laboratorium in cooperation with the Institute of Acoustics at the Norwegian Institute of Technology, Trondheim. The method is based on geometrical acoustics, *Cremer (1)*, and is a computerized version of the well known sound ray tracing technique. The calculations are done by the aid of a 3-dimensional mathematical description of the room. The programme developement has continued since 1967 and the present computer programme contains 18 separate parts. They permit the use of several thousand rays in the most complicated room configurations. It is thus possible to investigate sound energy distribution for many alternative room shapes for a given project and to compare the results with known room data (calculated and measured) for existing rooms, *Beranek (2)* and *Strøm (3)*.

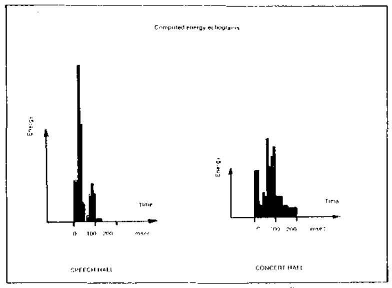


At any position within the room, in the audience area as well as on the stage, the output data are plotted to show:

- ☐ **the space distribution**
- ☐ **the time distribution**
- ☐ **the directional distribution**

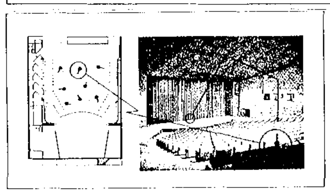
of early reflected sound energy.

Time — energy echograms at selected points can also be obtained. Experience from a variety of projects clearly indicates relationships between such echograms and the subjectively judged sound quality of different types of rooms. This has also been pointed out by *Schultz (4)*.



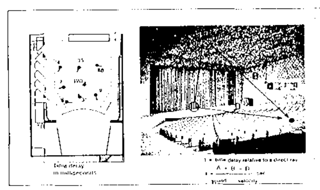
Apart from the study of echograms and the time — space distributions, the objective data evaluation is based on the use of well known acoustical parameters such as:

- ☐ «Schwerpunktzeit», *Kürer (5)*
- ☐ «Deutlichkeit», *Thiele (6)*
- ☐ «Directional factor» or «Spatial impression», *Schroeder (7)* and *Marshall (8)*



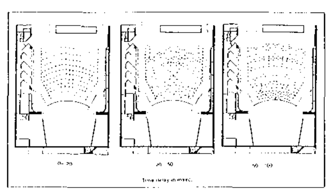
DATA OUT I:

The impact points for each ray that has hit the «microphone surface» will be plotted on a plan view of the surface. The direction of incidence is shown by a short line drawn out from the impact point. The length of the line (L) indicates the vertical incidence. L = O indicates a normal incidence on the surface, and L = x mm shows grazing incidence (x depends on the size of the drawing).



DATA OUT II:

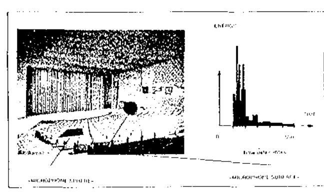
For each sound ray impact on a «microphone» the time delay is calculated. This means the additional time a ray will use when it is reflected at the room surfaces instead of going directly from the sound source to the impact point on the «microphone».



DATA OUT III:

Based on the calculated time delay for each impact, the impacts can be sorted into time delay intervals. This indicates the time distribution of reflections. The intervals can be choosen arbitrarly, but these intervals are often used:

0	msec	(direct sound)
0 — 20	msec.	(reflections)
20 — 50	msec.	
50 — 100	msec.	
100 — 200	msec.	
200 — 500	msec.	



DATA OUT IV:

Data registration on the «microphones» can be drawn in a time — energy histogram, with constant or variable time delay intervals. These histograms are called *energy echograms*. The same data can also be used to compute many room parameters: «Schwerpunktzeit» (t)[x], «Deutlichkeit» (D)[xx] etc.

x : t = centre of gravity of energy on the time axis
xx : D = the ratio between early incoming energy (0 — 50 msec.) and the total energy

REFERENCES:

(1) : Cremer, L.: «Die Wissenschaftlichen Grundlagen der Raumakustik Band I: Geometrische Raumakustik» Verlag S. Hirzel Stuttgart (1948).
(2) : Beranek, L.L.: «Music, Acoustics and Architecture» John Wiley & Son Inc New York (1962).
(3) : Strøm, S : «Distribution of reflected sound energy in models of existing concert halls». Technical report LBA 381 (1971). Akustisk Laboratorium — SINTEF. (In Norwegian).
(4) : Schultz, T.J.: «Acoustics of the Concert Halls». IEEE Spectrum 56 (1966, june).
(5) : Kürer, R.: «Zur Gewinnen von Einzahlkriterien bei Impulsmessungen in der Raumakustik». Acustica 21 (1969).
(6) : Thiele, R.: «Richtungsverteilung und Zeitfolge der Schallrückwürfe in Räumen». Acustica 3 (1953).
(7) : Schroeder M.R. et. al.: «Acoustical Measurements in Philharmonic Hall (New York)» JASA 40 (1966).
(8) : Marshall A.M.: «A Note on the Importance of Room Cross-section in Concert Halls» J Sound Vib. 5 (1967).

MASON INDUSTRIES

Mason Industries, Inc. is an outstanding source of vibration control devices. They excel in advising potential users of their equipment and optimum application techniques. They offer vibration isolators for applications as diverse as suspending a loudspeaker from a structure, with isolators available that include earthquake shock snubbers, to the isolation of entire rooms, if needed.

If you need vibration isolation devices, Mr. Mason has offered to send their very complete catalog to you upon request from you on your letterhead. Mason Industries, 92-10 182nd Place, Hollis, New York 11423

SYN-AUD-CON SPONSORS

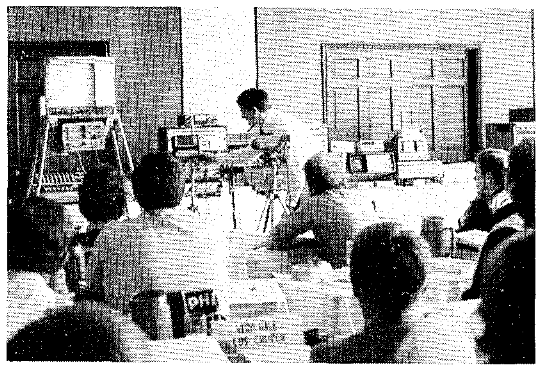
We started our Syn-Aud-Con sponsorship program in 1974 for two reasons. First, to try to keep our prices affordable by audio people, especially young people. Second, we wanted to have a working relationship with manufacturers so that we could share what we were learning from Syn-Aud-Con graduates. The program has been very successful on both counts. If it were not for Syn-Aud-Con sponsors, a Syn-Aud-Con seminar would cost over $500 this Fall; and we feel that we have a "family" relationship with 10 sponsors, one in which we give and receive ideas and information with 10 manufacturers in our audio industry. We feel that the entire audio industry has benefited from our sponsorship program.

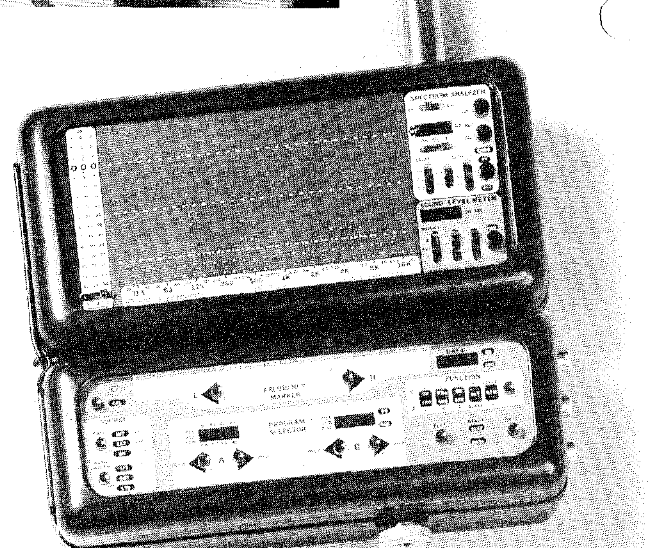
We have a "Syn-Aud-Con Sponsor Information" sheet which lists each of our sponsors, their products, their address and phone number, and your contact if you have a marketing or engineering item to discuss with them. We have included an up-dated sheet with this mailing so that you have it for your file.

IVIE

You don't need Syn-Aud-Con to tell you that Ivie is coming on fast. The IE-30A has achieved remarkable early acceptance in the real time analyzer market place. The only complaint heard today by Ivie distributors is that they can't get them fast enough. The IE-17A is next up.

During our Salt Lake City class we asked Brian Larsen, Vice President of Marketing at Ivie, if he would demonstrate the IE-17A to the class.





The system as shown in the photograph is, with the addition of an oscilloscope very close to being the complete audio systems measuring tool. Very minor additions such as a constant current output for impedance measurements, etc., are the type of detail modifications that might be made in the future.

After the Salt Lake City class we visited the new Ivie plant in Orem (just north of Provo, the home of BYU). We'll gladly award them the most scenic plant site in the United States -- Mountains to the East, a large lake and more mountains to the West, and a picture book fertile valley to the north and south. The view from the plant (high on a hillside) is spectacular. Now add a really modern, large facility with anechoic chambers, reverberation chambers, an exceptional roof top test area and you begin to get the picture. If you happen to be a skiing fan, or a shooter, hunter, horsebacker, hiker, camper, then Utah has to be an excellent prototype of Heaven.

We are really impressed by the environment Ray Ivie has chosen and enhanced, and by the very high caliber of men he has chosen to work with him.

MAY AES CONVENTION

We have again reserved Suite 1400 during the first three days of AES so that Syn-Aud-Con graduates can meet, rest and refresh during the AES convention. And, again we will have Booth 58 on the exhibit floor. See you May 15-18 in Los Angeles at the Hilton Hotel.


SALT LAKE CITY CLASS 1979

HAAS EFFECT

Recently in re-reading E. Madsen's paper, *The Disclosure of Hidden Information in Sound Recording*, I rediscovered his excellent illustration of the Haas Effect. (The paper was privately circulated but it may have been published later.)

it should be borne in mind that the curve shown is true of tests in an anechoic chamber. You will recall from class that 20 msecs works best in normal meeting rooms. In the write-up following, Kutruff's work points out that in the presence of a really "hot" first reflection followed by other reflections that "live" in the 20msec region, the Haas Effect can extend to 50 msec.

Quoting from Madsen's paper:

> Haas at the same time showed that within a certain time interval, two separate impulses immediately after one another are integrated by the brain to make the second impulse be heard as a part of the first. It has been shown that this integration process is subjectively received as an amplification of the first impulse. (See Fig. 1.)

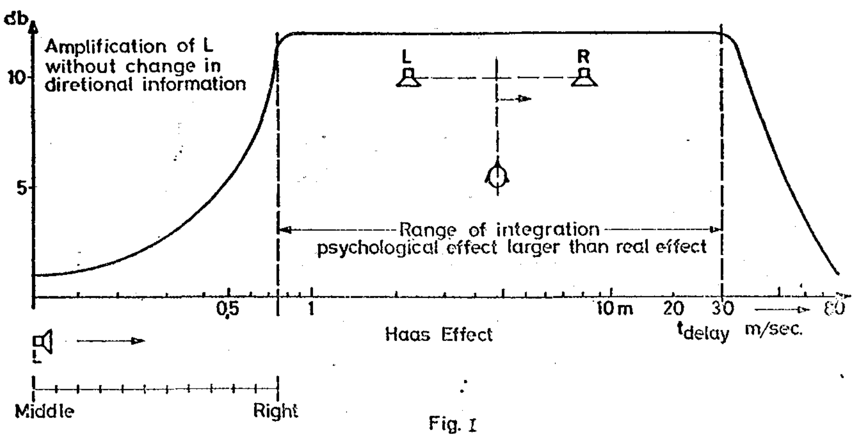


Fig. I

If a person sitting symmetrically between two identical pulse sources L and R receive exactly the same sound impulse from both sources, it appears to him that there is only one sound source midway between L and R.

But if the pulse L is increasingly delayed with respect to R, the sound picture moves quickly over to R, where it stays until the time delay goes over a certain value. Beyond this one hears two separate sound impulses. These conditions do not change, even if the source L is made louder than source R. It is only when the volume ratio exceeds values in the curve shown, that one recognizes L as a separate sound source.

The apparent volume of the sound source is subjectively felt to be greater than the actual volume of the source.

MORE ON LEDE - FROM TED UZZLE

The inimitable, indefatigable, and intuitive TED UZZLE has done it again. Ted's ability to find original papers is uncanny. The following arrived with the usual form of address: Dear Sir:

> There may be additional phenomena involved in live end - dead end room design than have been described so far. One such is dealt with at pages 169 - 170 of Room Acoustics by Heinrich Kuttruff (London, Applied Science Publishers, 1973).
>
> Kuttruff describes the familiar Haas effect, a masking (or high threshold of audibility) for about 20 ms after a direct sound. However, Kuttruff found something more. If there actually is a hot, hard reflection within the first 20 ms, the masked shadow zone will be extended beyond the 20 ms value.
>
> Please refer to the enclosed copy of the figure. When there is a reflection at 16.7 ms, this renews and extends the threshold of audibility to 30 ms (beyond which the Haas effect no longer controls, and the threshold level declines). When there is a reflection at 16.7 ms and also at 33.3 ms, the knee of the curve does not come until 50 ms.

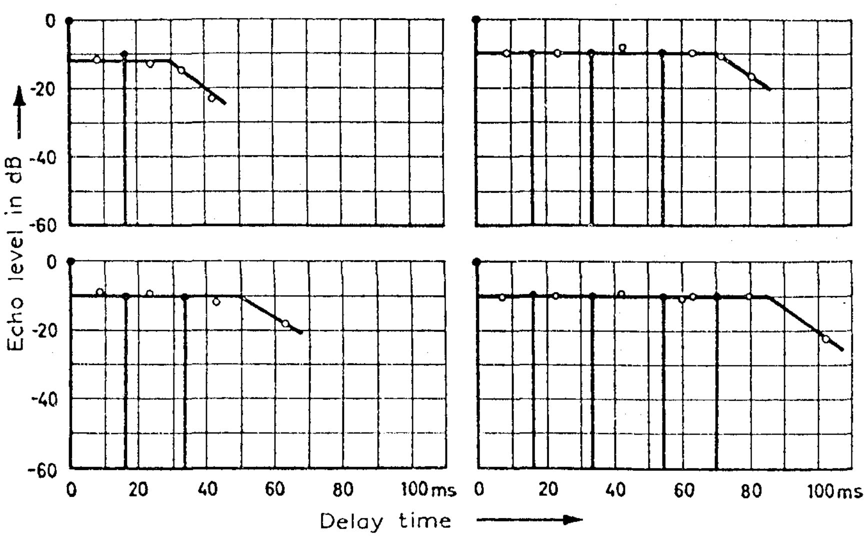
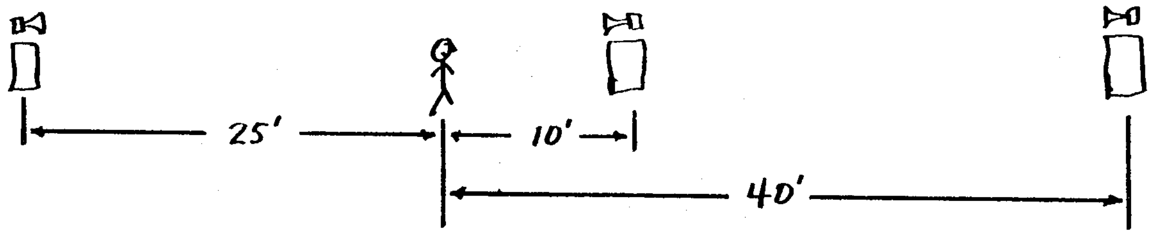


Fig. VII.3. Absolute threshold of perceptibility of a delayed signal (reflection) being added to a sound field consisting of direct sound plus one, two, three or four reflections at fixed delay times and relative levels, which are denoted by vertical lines. The original sound signal is speech. All sound components arriving from the front.

> Kuttruff says in the text that this applies only to reflections in the median plane of the head -- in front, from above or behind. The ears behave quite differently with lateralized reflections.
>
> We could possibly apply this to live end - dead end design thus. Some increment of the strong rear-wall reflections will travel beyond the mixer's head back into the dead front end, and then be reflected back to the mixer from the front again. Yet the very passage of this wave by the mixer from behind him will serve to renew and extend the temporal masking, and thus reduce the perceptibility of the second-order reflection. Thus the live end "helps out" the dead end.

Ted Uzzle, continued

Pretending that we can dampen a surface so completely that nothing is reflected back is like pretending we don't know where babies come from. There absolutely are going to be second and later-order reflections, no matter how carefully we deaden the front end. The very adroitness of the live end - dead end technique lies in the way we extend the "20 ms temporal masking effect" beyond 20 ms.



You can test this in the open air with an observer and three loudspeakers whose levels can be separately adjusted. One loudspeaker should be ten feet in front of the observer, one twenty-five feet behind, and one forty feet in front. Playing the same material through all three speakers, at what level does the signal from the rear speaker become separately audible? At what level does the distant front speaker become separately audible, with the rear speaker on? With it off? This would test the hypothesis quite specifically. Doubtless you can think of other, more convenient ways, using delay equipment. (Ed's Note: We will have a report on this next Newsletter. And anyone else performing the tests should write it up for the Newsletter.)

In addition to the above, Ted also called our attention to a paper in the February 1979 JASA Volume 65 #2, pp 524-527, *Effects of Early Multiple Reflections on Subjective Preference Judgements of Music Sound Fields*, Letters to the Editor Section, wherein among other fascinating data are statements, "This is further indication of the advantage of early lateral rather than non-lateral reflections" and "The delay time of the first reflection is not as important as the delay time of the strongest reflection".

Guess who can clearly measure and adjust that. ETC can!

TEMPERED SCALE

DAVID ELLIS, Manager of Engineering at Ohio Bell, wrote the following on the Tempered Scale. If it is a subject that interests you, be sure to read Hunt's *ORIGIN IN ACOUSTICS* reviewed in "Books of Interest" in this Newsletter.

While reviewing Newsletters in preparation for my second Syn-Aud-Con seminar, I found an article in Volume 3, Number 3, page 14 by John Freitag on the "Tempered Scale" that struck a nerve. A "part time" musicologist, I decided to look into this subject further and thought I would share my trivia with Syn-Aud-Con.

The relationship of science to musical sound is truly a which came first the "chicken or the egg" question. I would imagine that when the first caveman blew into a hollow branch and got a sound that another caveman designated himself "acoustical engineer" and began looking for more hollow branches. Whatever happened, the story is meager in historical facts until the time of Pythagoras, that great mathematician who evidently was interested in more than triangles. He applied some mathematics to the musical scale in about 550 B.C. that held until about the 16th Century with a minor variation sometime in between.

Pythagoras surmised that the scale (by that time it contained five whole tones and two half tones forming an 8 note scale or octave) was derived from the interval of the fifth to the original. A mathematical ratio of 3/2. Thus the third is the fourth successive fifth reduced by two octaves or 3/2x3/2x3/2x3/2x1/2/1/2 = 81/64. The fourth is 729/512 etc.

The Table shows all of the ratios as Pythagoras saw it. In a modification of this theory a system of "just" tuning or temperament was developed. It was based on the fifth being a 3/2 ratio as Pythagoras had it, but also included the third as a 5/4 ratio and thus defined all other notes in relationship to the 3rd or 5th. But even though the mathematics looked good, the musicians were not crazy about it. The "C" major scale had two different size of whole tone intervals, making modulation impossible. The "G" major scale had different intervals than the "C" major scale, hence two different tones of "A" would be necessary; one for the sixth of C, the other for the second of G.

MUSICAL SCALES

	Pythagorean Temperament		Just Temperament		Equal Temperament	
	Ratio	Freq (Hz)	Ratio	Freq (Hz)	Ratio	Freq (Hz)
C'	1	260.66	1	263.47	1.000	261.63
D'	9/8	293.24	9/8	296.40	1.122	293.66
E'	31/64	331.03	5/4	329.33	1.260	329.63
F'	729/512	371 17	4/3	350.42	1.335	349.23
G'	3/2	390.99	3/2	395.21	1.498	392.00
*A'	27/16	440.00	5/3	440.00	1.682	440.00
B'	243/128	494.73	15/8	495.32	1.888	493.88
C'	2	521.32	2	526.94	2.000	523.25

NOTE: All scales are based on 440 Hz - "A" for comparison purposc.

In the 18th Century the "even" or "equal" tempered scale was developed. It was decided that all whole tones would be equal intervals and half tones would truly be 1/2 of the whole tone value. Since the scale was defined as 5 whole intervals and 2 half tone intervals this gave us 12 half tone intervals to the scale also known as the chromatic scale. The half tone intervals was divided into 100 parts called cents. Therefore, there are 1200 cents to the modern scale. And who said modern music is not worth 2 cents! So the result can be seen in the Table is that in most cases the "just" or "pythagorean" scales are higher than the equal temperament. Which again proves that we musicians are not out of tune, we are just in a different temperament, which is to be sure, the derivation of the term "Temperamental Performers".

FOUNDER AT CROWN INT'L PASSES

Clarence C. Moore, founder and President of Crown International, Inc., passed away on January 24, 1979, at the age of 74.

A man's life is not measured by solar years but by the full exercise of his God-given gifts. The influence of Mr. Moore's life touched literally millions of other human beings on this planet and more importantly, touched them with goodness.

Inventive, entrepreneur, a profitable business man, a minister of the gospel, a missionary, a father, a teacher, and an industry pioneer, his path through life strewn with victories over difficult obstacles, he gave that ultimate proof of a man in contact with his God by living an uplifted life. Carolyn and I had lunch with Mr. Moore when he was in his early seventies and his enthusiasm drive, but most of all, his integrity generated that aura that wipes out all sense of age, time, or sense of self and left us with a precious memory that yes, business can be done that way.

Mr. Moore had, over the years, prepared Crown to grow and prosper without need of his immediate physical presence. His choice of strong managers indicated his own worth. It's been truly said, First rate men choose first rate men to work for and with them. Second rate men choose third rate men.

COMMENTS FROM DAVE KLEPPER

We recently received a letter from DAVE KLEPPER, the well-known and respected acoustical consultant who is one of the principals of KMK Associates in White Plains, New York. Dave has a few comments on our January 1979 Newsletter. We reproduce the letter almost in its entirety with some editorial comments. Dave Klepper, is in our opinion, the most widely experienced designer of sound systems active in our field today and it is always informative to receive his thoughts regarding these important subjects.

Dear Don:

This letter is prompted by the most recent material received from you. As usual, the Newsletter and Tech Topics are valuable to us, but several items prompt some discussion.

A. Newsletter Volume 6, Number 2, Page 26, "Books of Interest":

I cannot help be happy that you gave the AES Anthology a "plug". I know full well that you are not a person who claims credit for other's work, but your casual reader who then buys the Anthology may wonder just what Pete Tappen and I did, after you selected forty-five of the papers printed. What you probably meant, I am sure, is that you agreed with Pete and me on these 45 papers (Ed's Note: Dave's surmise is correct) and do not want to be responsible for the selection of the additional papers, inclusion of which you didn't agree with. Obviously, many or most of the papers you selected were also selected by Pete and I, and this should be brought to the attention of your readers. Otherwise, it appears that Pete and I are taking credit for your work! (Ed's Note: Everything we write doesn't always come out the way we intend. We often have only a week for the writing, typing and paste-up of the quarterly mailing, which doesn't leave us a lot of time to think through each word and its meaning. The AES Anthology review is a beautiful example of the problems we generate in our haste.)

B. Newsletter Volume 6, Number 2, Page 16, "Ideal Polar Pattern":

Your description suggests that horns of the wrong coverage pattern were tried, and that a combination of two Altec Manta Ray or two Electro-Voice high-Q horns would have solved the problem much better than any single horn. I am disturbed that so many designers of central clusters still try to "get by" without using the light-analogy technique discussed by Wilfred Malmlund and Ewart Wetherill in 1964 and now reprinted in the above AES Anthology. (Ed's Note: The light analogy technique is excellent for determining coverage angles but can lead to an excessive N for the Q's available if careful attention is not devoted to the calculation of articulation losses.) While Richard Negus doesn't use this technique, and his results are successful, he generally uses more components than I, or Pete Tappen, or Jacek Figwer (to name a few who do use the light-analogy technique) would in a similar situation. Others, who don't use the technique usually try to get by with one or two horns when more are required, and coverage suffers in consequence.

We have used both Electro-Voice High-Q and Altec Manta-Ray horns on several projects. We have found that using them as the manufacturers would have you use them is usually successful, but there are important exceptions. In a theatre used for a Broadway musical show, the balance between the usually live pit orchestra and the electronically-reinforced singers must be preserved in all seats to avoid complaints. Relying on coverage at the "ragged edge", where it is 6 dB down, will not maintain this balance. So, we now usually design with more overlap than either Altec or E-V would suggest, and consider a 40x60 horn to really provide 35x50 coverage, a 40x90 horn to really provide 35x80 coverage, and a 20x40 to really provide 15x35 coverage. A study of the actual polar patterns at various frequencies can help design a system more accurately. (Ed's Note: What Dave has to say here is valuable and we know from experience the excellent coverage he provides in his systems. What we were objecting to originally is the inability to *vary* the main pattern in level for front-to-back seats.)

C. Tech Topic Volume 6, Number 5, "The Sabine Reverberation Equation..."

This otherwise excellent reprint still persists in the error that the way to calculate audience and seat absorption is still on a per-person and per-seat area, whereas Beranek's work in Music Acoustics and Architecture, and his "Audience Absorption in Large Halls, 11, JASA, January 1969, points out to anyone who will read, listen, and give it a try, that calculations on a per-area basis are much more accurate, for large halls anyway. We generally use the per-area method and the coefficients in the 1969 JASA paper, and we find that measured results meet predictions much better than through the use of the per-person or per-seat techniques. Use of the per-person and per-seat techniques is not really feasible for accuracy, anyway, because of the wide range of coefficients given. (Ed's Note: It is our feeling that what Dave is saying is correct, but for *simple* cases of *electroacoustic* design such accuracy may not be needed.)

I hope you will bring these matters to the attention of your readers. Keep up the good work.

SYNERGETIC AUDIO CONCEPTS

"THE TWENTY"

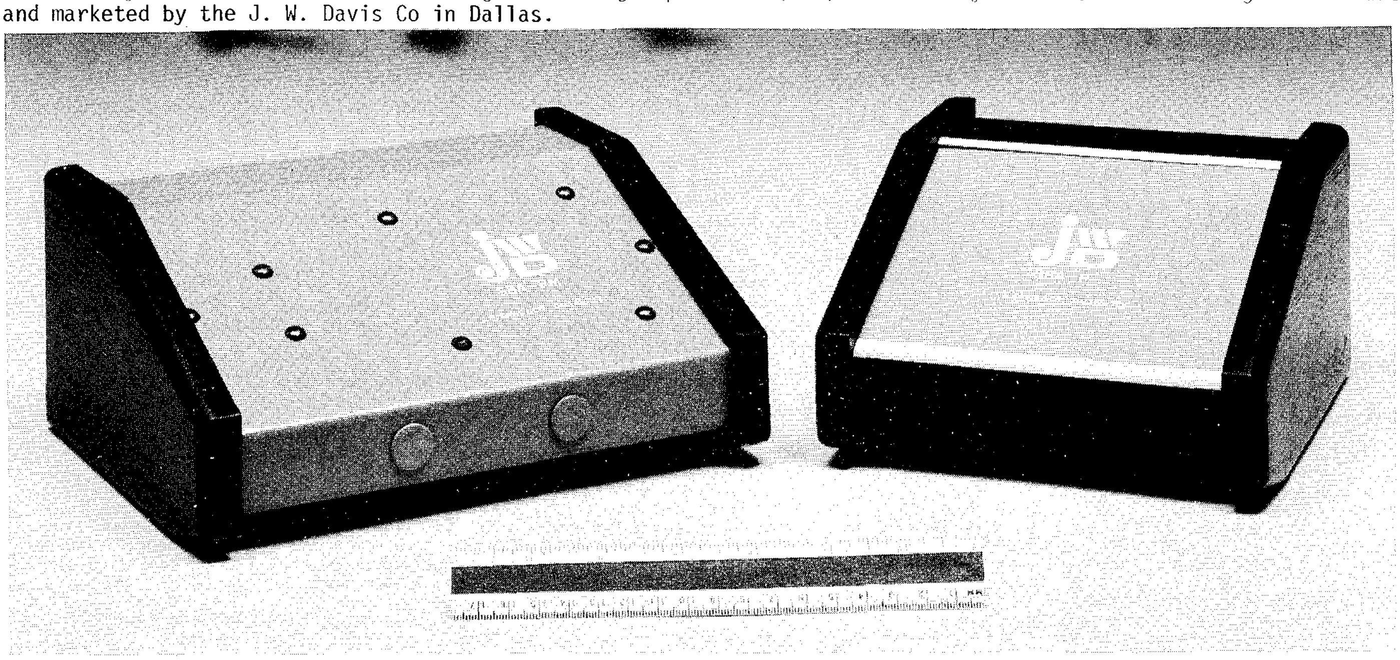
Within ten days of the first announcement of the special TDS Workshop, May 9-11 at the Filmways-Heider Studio, it was fully subscribed. "The Twenty" certainly represent the quick and the dedicated. They are in for a unique experience. Just take a look at the ETC technique elsewhere in this Newsletter for a small sample of what they will have *hands on experience* with. TDS licensees have now crossed the 100 mark and continue to grow weekly.

We are particularly pleased to discover that there are twenty pioneers ready for really advanced work in audio. There are thousands of real time analyzers in use today just ten years after its first use in audio by us in 1968. It took several years to get 100 users of RTAs. We'll be very surprised if TDS-ETC doesn't see thousands of users by 1990.

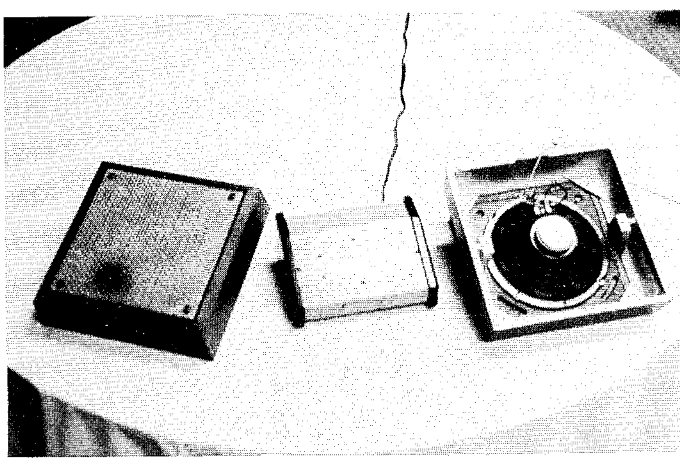
What the twenty are going to witness and participate in is a hands-on look at just what might be readily available tens years from now. When there is dedicated TDS apparatus, and FFT units are built to interface specially with them, then what we are using now will seem clumsy by comparison. Right now its ten years ahead of the crowd and so are the twenty.

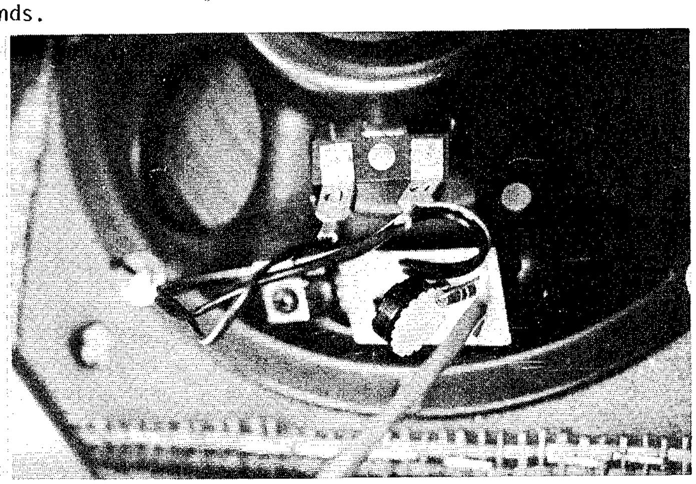
SBA - PROTOTYPE BEING TESTED NOW

The January Newsletter announced the Signal Biasing Amplifiers (SBA) invented by Dick Heyser and being manufactured and marketed by the J. W. Davis Co in Dallas.



Here are photographs of the prototype units we have been demonstrating in Syn-Aud-Con classes in 1979. The SBA-PM is the master unit with a builtin power supply for the first twenty or so remote units. The SBA-P is an additional power supply to be used as the system expands.





The remote units mount right to the loudspeaker frame and use it for a "heat sink". A typical 8" loudspeaker and baffle is shown with the SBA-PM in between to illustrate the scale. The close up of the remote unit shows it to be very simple board screwed to the loudspeaker frame.

In demonstrating these units we have found them to be rugged and reliable. You will be hearing much more about SBA. The present units are now in the field being tested. If you want literature and pricing on the SBA units, write the J. W. Davis Co., 9212 Denton Dr., Dallas, TX 75235. If you want to call, talk to Chappie, 214-352-8405.

"COLLAPSABLE SPEAKER ENCLOSURES"

Once in a while one has to go through old files. And sometimes old files contain gems. We have reproduced here 1 of 3 pages from a "26 year old architecture student at Stanford University" and Don's reply. And I think the picture of Don sort of goes with the text.

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PRESIDENT
~~Sales Manager~~
Altec Lansing
1515 South Manchester Avenue
Anaheim, California

Dear Sir:

I have a great idea. It is a project that may profit your company, many high-fidelity enthusiasts, and me. My project is to build collapsable speaker enclosures following the construction drawings of your VOICE OF THE THEATER as presented by Alexis Badmaieff and Don Davis in their book HOW TO BUILD SPEAKER ENCLOSURES. Concurrently I will write an article on my project and submit it to be published nationally.

For a long time I have admired your VOICE OF THE THEATER speakers. For their full-bodied performance, your price looked small enough to handle. Presently they were too large, but one day when I would have a permanent residence I would buy them. I thought in this stereotypic line until I figured that it was not their size but their immovability, their non-portability, that deterred me. If the enclosures could be dismantled they would be moveable. It is my opinion that collapsability can be a key feature for many people who yearn to own full-size speakers. People are thrilled when they hear full-size speakers, but they are dismayed and worried by the fact that moving would be a serious inconvenience. Moving collapsable speakers would be a one-man job; parts could be hauled away on a regular car rather than a moving van. People do not move very often, but people do have a deep fear of being immobile. Collapsability would be a welcome feature because it allows people to think that they can pick up and go whereever and whenever they want to or need to.

The above may be pure theory, but a sales boost of Altec speaker components immediately following my article's publication will prove the soundness of my theory in hard figures, in which case, you would be wise to market a third model--the collapsable VOICE OF THE THEATER--in addition to your deluxe walnut model and your utility model.

ALTEC LANSING
A DIVISION OF LTV LING ALTEC, INC
1515 SOUTH MANCHESTER AVENUE, ANAHEIM, CALIFORNIA · 92803
774-2900 AREA CODE 714

September 23, 1969

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Dear Mr.

Thank you for your kind remarks concerning "How to Build Loudspeaker Enclosures".

Actually Alex and I developed such a collapsible loudspeaker about 15 years ago. Unfortunately, one of the prototypes was stolen, while collapsed, and sold to some unsuspecting Easterners. Not realizing that it was a collapsible A-7 and lacking the aural acuity to recognize that "the full bodied performance" of a true A-7 was missing, they formed a combine and marketed the first acoustic suspension loudspeaker.

Actually, what happened, the prototype collapsed and smashed the cone driver, giving it a considerably greater travel. Not only did we never receive recompense for our prototype, but the Eastern combine failed to include our names on their patent application.

This tragedy stunned both of us so severely that we failed even to publish our earlier findings.

I hope that your own research is blessed with happier fruition. I'm torn between approaching the problem by making an expandable shoe box model or the earlier approach of a collapsible large loudspeaker. While Alex and I often theoretically discussed an inflatable loudspeaker that would swell and recede in proportion to the dynamic range of the program material we again failed to protect ourselves and now we find the automobile manufacturers installing them as safety devices in the 1971 model cars.

I wish you considerable more success than Alex and I have had.

Sincerely,

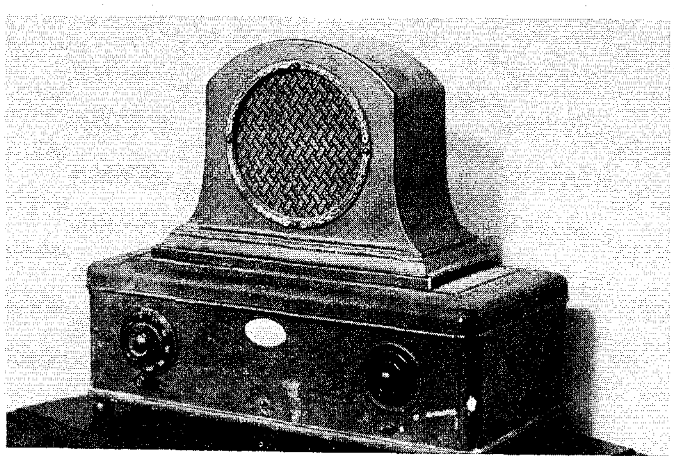
Don Davis

Don Davis

DD:c

DEWARD TIMOTHY'S ATWATER KENT RADIO

DEWARD TIMOTHY of Salt Lake City, a graduate of three previous Syn-Aud-Con classes, helped us do our 100th class in Salt Lake City (our first there). Deward owns and operates Poll Sound, having purchased it from its founder who goes back into the 1920s with his business. Deward and his wife, Kathy, were two of the many people in Salt Lake who shared their time with us and made our stay especially beautiful. Deward showed us a few of the relics from Mr. Poll's early days in audio. And he presented us with an Atwater Kent radio of the 1927-1928 vintage *that actually works*.



What is especially pleasing to me is that my first glimpse into the world of electronics was an Atwater Kent receiver of exactly the same vintage. When I was about 12 years old I lugged one 6 miles. I collected this model of Atwater Kent and built my first ham radio transmitter into one of the metal cases just before World War II.

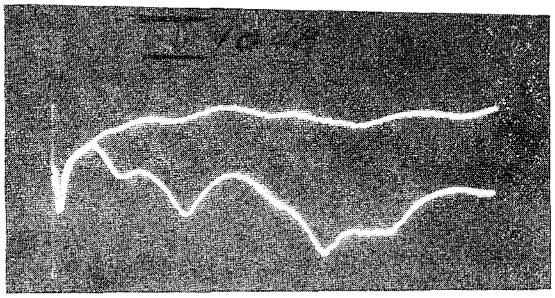
The large volume control on the right hand side of the front panel when removed left a hole just the right size for a plate current meter in a small cw transmitter (remember 6L6s?).

When we got home from SLC, I dug out an old photograph of my ham rig with my first small transmitter in an Atwater Kent cabinet with the Hallicrafters S-20R receiver sitting on top of it.

SAN FRANCISCO CLASS 1979

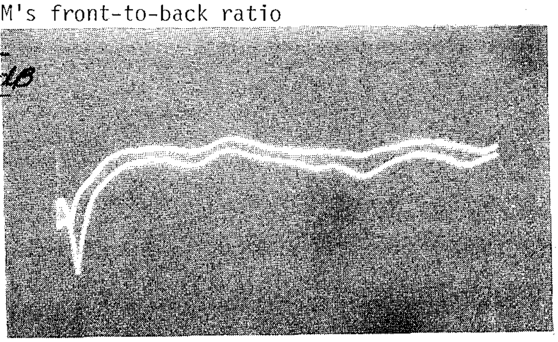
MAKING THE PZM™ INTO A DIRECTIONAL MICROPHONE

The installation of a PZM near an orchestra pit, a drum set, or other high level source tending to "mask" the desired signal such as a vocal solo can be handled with a small piece of good quality carpet. (I'm sure there are other good isolators however the carpet is what we had at hand when we made the test in class and recorded the measurements.)

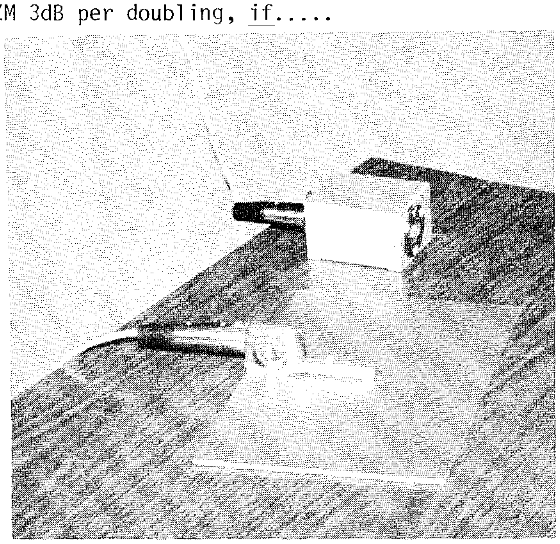


PZM's front-to-back ratio

10dB



PZM 3dB per doubling, if.....



The photograph is of the PZM's frontal response (top curve) and its rear response to the same source (bottom curve). 20 to 30 dB is typical.

This, of course, is one of the *basic* advantages of PZM, namely, that the presence of something like a carpet does not cause undesired anomolies of its own.

The second photo reveals another property of PZMs and that is that it follows 3dB per doubling of distance IF, and its a big IF. ROGER ANDERSON of Shure Brothers pointed out in the San Diego class that this can only occur if the source also is located on the boundary surface.

Every class leads to new and fascinating insights into PZM techniques and technology. The letter reproduced below is one of the more conservative ones. We would like to hear from you. What have been your experience with the PZMs? What do you find to be it's major advantages? Disadvantages?

SPELLBOUND SOUND
2449 FIRST STREET
FORT MYERS, FLORIDA 33901
813-334-0594

MARCH 21ST, 1979

MR. KEN WAHRENBROCK
WAHRENBROCK SOUND ASSOCIATES
9609 CHEDDAR STREET
DOWNEY, CALIFORNIA 90242

DEAR MR. WAHRENBROCK;

I AM TAKING THIS OPPORTUNITY TO REPORT TO YOU ON OUR EXPERIENCES WITH THE PZM.

1) AS HAVE OTHERS, WE HAVE HAD TERRIFIC SUCESS USING THE PZM FOR PICKUP OF THE PIANO. ACCORDING TO THE PLACEMENT TECHNIQUE USED, WE ARE ABLE TO GET A MULTITUDE OF NATURAL PIANO SOUNDS WHICH ARE AGREEABLE TO (AND PREFERRED BY) ANY LISTENER, WITHOUT THE NEED FOR ANY EQUALIZATION WHATSOEVER. THE DEGREE OF SUCCESS IN OBTAINING THE DESIRED SOUND (SPECTRUM) IS DIRECTLY PROPORTIONAL TO THE NUMBER OF DIFFERENT PLACEMENTS TRIED.

2) WE HAVE BEEN USING THE PZM AS A MEASUREMENT MICROPHONE, AND HAVE BECOME AWARE OF IT'S USE IN SOLVING ACOUSTICAL PROBLEMS. AS AN EXAMPLE: A PARTICULAR CHURCH SANCTUARY HAD A MEASURED RT(60) OF APPROXIMATELY 4.0 SECONDS. PLACEMENT OF THE PZM AGAINST THE REAR WALL (USING THE PZM AS THE MEASURING MICROPHONE FOR THIS TEST) YIELDED AN RT(60) OF 2.2 SECONDS. IT WAS THEREFORE DEDUCED THAT THE REAR WALL WAS ADDING 1.8 SECONDS OF RT(60) TO THE ROOM. FROM CALCULATIONS PERFORMED IT WAS FOUND THAT IF THE REAR WALL HAD BEEN ABSENT (OR 100% ABSORBTIVE), THE RT(60) OF THE ROOM WOULD HAVE BEEN INDEED 2.2 SECONDS. THEREFORE IT IS MY CONCLUSION THAT THE PZM WAS NOT "SEEING" THE REFLECTION OF THE SOUND FROM THIS WALL IN THIS TEST. I BELIEVE THAT THIS POINT MAY BE FAR REACHING, AS WHEN APPLIED TO PRACTICE WILL AID US IN IDENTIFYING AND QUANIFYING PROBLEM SURFACES, MORE PRECISE ALLIGNMENT OF LOUDSPEAKERS, AND MUCH MORE. I WILL PROVIDE YOU WITH MORE PRECISE INFORMATION ON THIS TEST, AND ANY OTHER TESTS WE PERFORM, AT YOUR REQUEST.

3) WE ARE STUDYING THE EFFECTS OF MOUNTING THE PZM DIRECTLY TO SURFACES OF VARIOUS TYPES OF MUSICAL INSTRUMENTS (GUITARS, DRUMS, WIND AND BRASS INSTRUMENTS) AND WILL LET YOU KNOW THE RESULTS AS WE EXPERIMENT.

IN CLOSING, I MIGHT EXPECT THAT THE NEXT FEW YEARS MAY SHOW US THAT THE PZM MAY BE ONE OF THE MOST IMPORTANT DEVELOPMENTS IN THE AUDIO FIELD, IN THIS DECADE.

BEST REGARDS,

MARTIN TOWNE - MANAGER

GROUP DELAY DISTORTION

One loudspeaker manufacturer, faced with several seriously time misaligned loudspeakers in their product line, have evoked an Acoustical Society of America paper by J. Blauert and P. Laws (West Germans), entitled *Group Delay Distortions in Electroacoustical Systems*, May 1978, pp 1478-1483. Interestingly enough, the manufacturer's premier product fails to pass Blauert and Laws criteria, as well.

Be that as it may, the real point, we feel, is that group delay distortion has to be a minor problem. What all of these investigators seem to miss and certainly the loudspeaker designers have totally missed is that minute time differences measured in microseconds cause amplitude response anomolies measured in octaves. Blauert and Laws used a bandlimited impulse signal and therefore effectively eliminated consideration of the amplitude problems that can appear elsewhere in a full spectrum system.

Having lived some months now with correctly time aligned[tm] systems we are convinced in our own minds that quite small discontinuities in time can be detected when they occur abruptly, as they do near crossover, for example, though it may well be the amplitude problems they cause and not necessarily the time delay itself that we actually detect. The Blauert and Laws paper is an interesting one and their criteria would serve, in our opinion, as a rough first cut at *physical* alignment of drivers. It fails, however, to explore the really subtle area under 250 usec between drivers covering the same frequency in the crossover region.

Dick Heyser assures us that he too *easily* hears very narrow amplitude response anomolies well inside critical bandwidths and that just a small exposure to them, if the listener is mentally alert to their presence, quickly trains the ear to recognize them whenever they are present. (We have been told for years that the ear can't sort out information inside critical bands. But this is similar to saying that we can't hear differences of less than 3dB. We have been demonstrating in class recently that most of the class can hear 1/2 dB changes and we are assured that trained ears within the recording industry can hear 1/10th dB changes, yet an untrained ear hears only 3dB difference.)

All of this, perhaps, points up the fact that many a strongly expressed preference for some loudspeaker's "sound" is really a vote by the listener for the distortion he has come to know and love.

ARE YOU CORRECTLY POLARIZED?

The influence of absolute polarity is again receiving serious attention among listeners with (a) good ears, (b) good systems, (c) good sources.

In recent classes we have included polarity switches on each of the systems we use. Our speech reinforcement system (single channel) shows a definite sensitivity as to total polarity. Perhaps the easiest way to share our experiences with you is to describe three cases.

Case # 1

We were comparing two different +$1,000 amplifiers in a recording control room (UREI-813A monitors). Everyone had a definite preference for one of the amplifiers *until* we switched the polarity on *both channels* of the second choice amplifier. Then they were extremely difficult to tell apart. We couldn't help wondering how many A-B tests have been invalidated due to this effect.

Case # 2

We were demonstrating polarity reversal on a single channel speech system and everyone was able to hear the difference easily *until* I stepped out of the room with the wireless microphone on. Then reversing polarity made very little *detectable* difference. Obviously the non-symmetry of the speech waveform made the comparison of the reinforced speech vs the live speech easily heard.

Case # 3

We were equalizing the same system with great difficulty at another class location -- violent feedbacks closely spaced, etc. Reversing polarity made the tuning easy and removed feedbacks that seemed to have a "trigger level" when they came on. Suddenly, we realized that we were using a *different* wireless microphone system. Upon checking we found that the Swintek had one polarity out of the receiver and the HME was the opposite.

John Hilliard states he observed and corrected for this in theater sound systems as early as 1937. Heyser and Long both stress the importance of total polarity being correct.

It is good practice to put polarity switches on each channel and try both ways before doing any real serious listening. You shouldn't be surprised to discover that you will need total polarity reversals from record to record. Let's hope that one day the whole chain from orchestra to loudspeaker is clearly specified as to polarity.

MORE ON M_E

In Tech Topic Vol 6 # 4 the statement is made that

$$M_e = \frac{N\,\Omega_{min}}{\Omega_{avail}}$$

JOHN CARTER, engineer at Bose, suggested in our New York class that

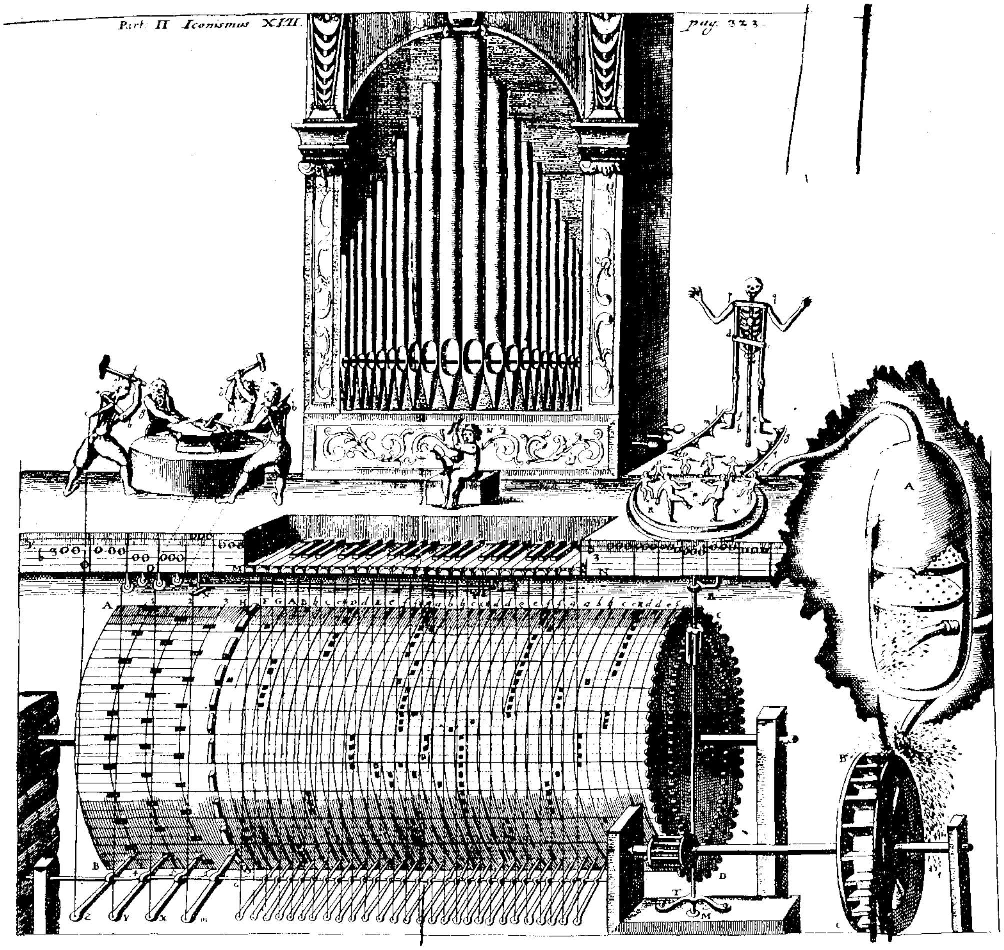
$$N \approx \frac{\Omega_{min}}{\Omega_{avail}}$$

When the answer is a decimal fraction it is rounded up to the next largest number. This is because

$$\frac{\Omega_{min}}{\Omega_{avail}} = \frac{M_e}{N}$$

When $\frac{M_e}{N} = 1$, you have a workable situation in terms of required shortening of D_2. Thus, an N that insures that $\frac{M_e}{N}$ is equal to or less than one (1) is a usable number

MAGIAE UNIVERSALIS



A water-powered automated musical instrument that so interested our forefathers, as the computer does today.

During our Fall 1978 class tour in the East we stopped over in Williamsburg, Va. As is our habit we explored the local old-book stores. Upon asking if they had any books on acoustics we were told they had just one - an old one. What they brought out for our inspection was a Vellum bound volume, first edition, dated 1657 entitled *MAGIAE UNIVERSALIS*, written by Gaspare P. Schotto. The book is written in Latin and was published by Herbipoli.

The frontispiece is reproduced here.

In researching the background of this book we find that it is mentioned in the Edinburgh Magazine, Volume 12th, page 322 for the year 1790 and more recently in Hunt's *Origin of Acoustics* where it is referred to in regard to Boyle and Hooke's work.

In fact, it was this book that gave the description of Otto Von Guericke's work with air pumps. Kasper Schott was a contemporary of Von Guericke's. Gaspare P. Schotto (the Latinized version of his name) was used by Robert Boyle (1627-1691) and his assistant, Robert Hooke (1635-1703) in their vast improvement in air pumps and their subsequent experimentation with ticking watches in vacuums.

There are some truly remarkable illustrations in this book and we will plan to reproduce some of them in future Newsletters, including excellent isometric renderings of water powered automated organs and other musical instruments. The unique programming charts for these mechanical wonders are included.

Kasper Schott was a Jesuit professor of Physics at Wurzburg (1608-1666) and his pragmatica, corollarium, annotatio, provemium, etc. are fascinating. There are even plots of Diatonica, Chrematica, and Enharmonica.

This venerable volume is now further distinguished by being the earliest source book in Syn-Aud-Con's collection. Dick Heyser put it well when he remarked, "You don't really own that book -- you are its temporary custodian". Amen!

MAGIAE UNIVERSALIS, continued

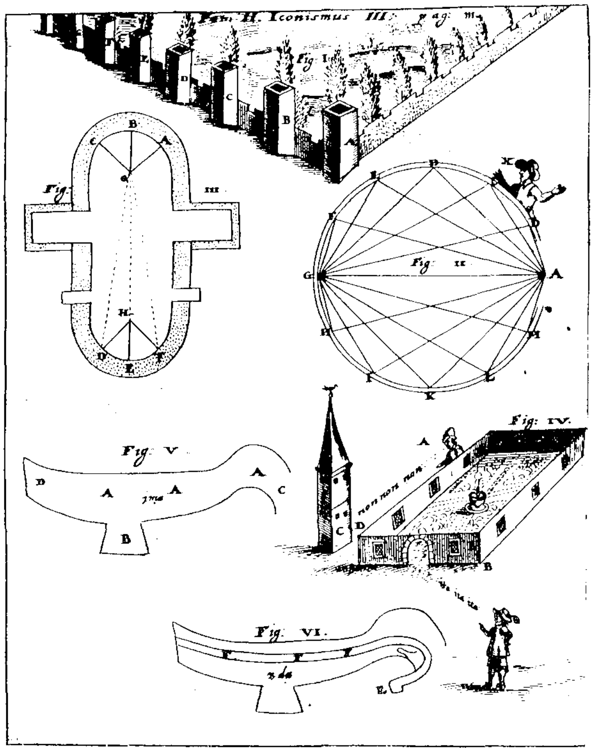


Fig. # 1. Reflection, focusing, diffusion, time delay, creeping are all illustrated here.

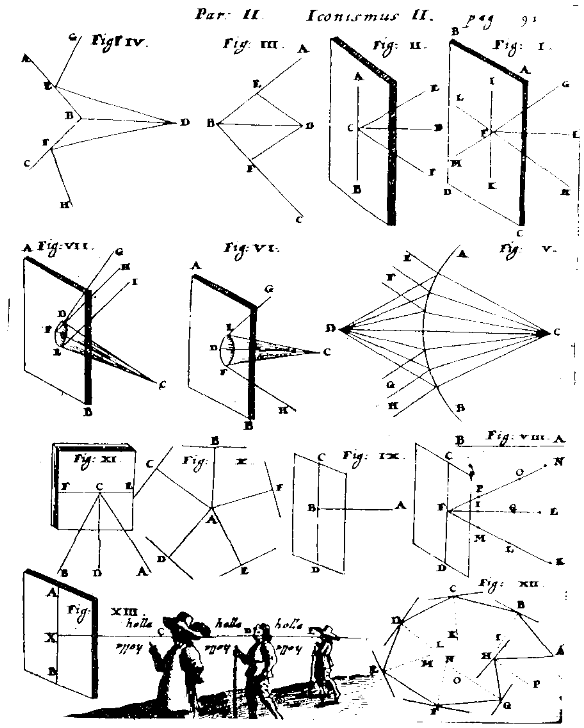


Figure # 2. The basic rules for sound reflection as a geometric problem.

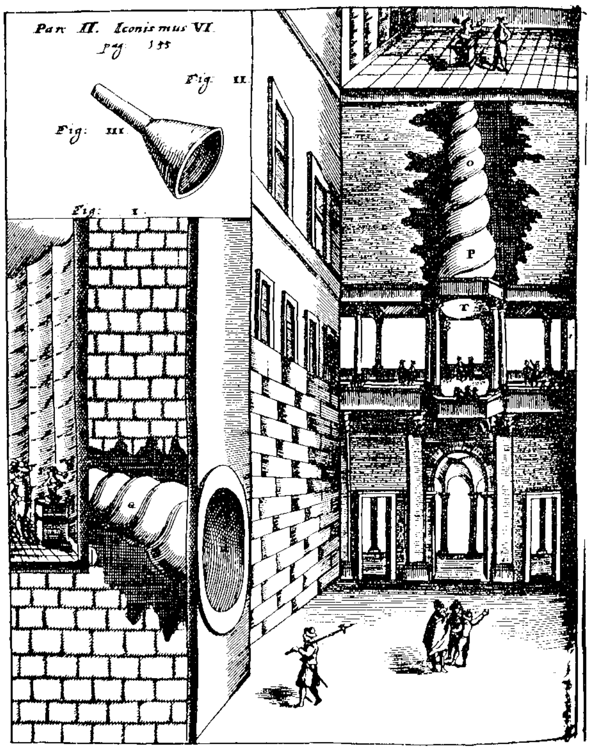


Fig. # 3. How "oracles" talk or music can be "transmitted" from one space to another.

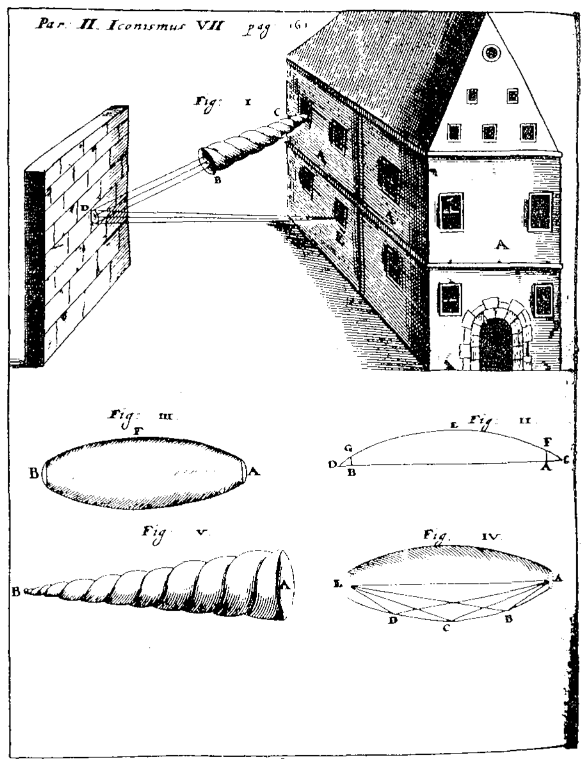
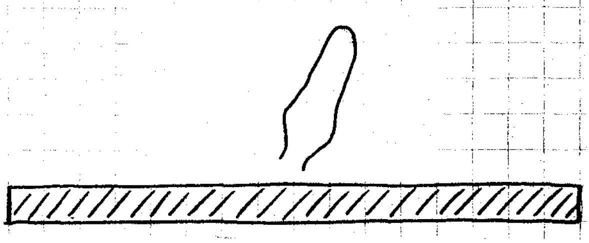


Fig. # 4. The first "bugging" system. Such horns were actually used by Athanasius Kircher, a contemporary of Kasper Schott, to speak to the gate keeper from his quarters and to eavesdrop on the conversations taking place in the courtyard. His experimental horn was 22 palma long (If you assume a palm as 8.7 inches, then his horn was about 16 feet long. -- Ted Hunt in his book, Origin of Acoustics sets the length of the palm at 8.7 inches.)

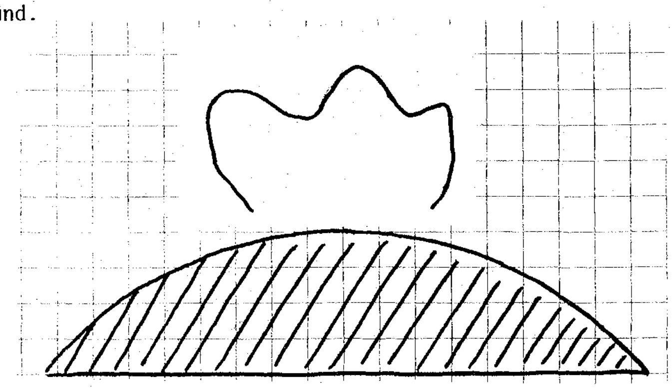
DIFFUSION

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

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



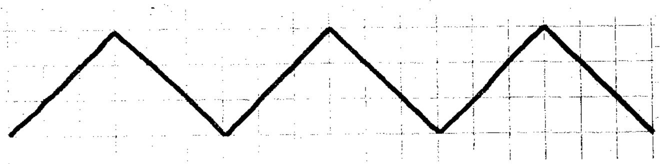
1a. Flat Surface Reflection



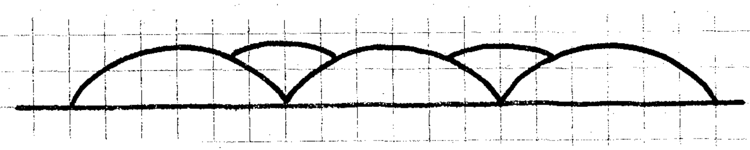
1b. Cylindrical Surface Reflection

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

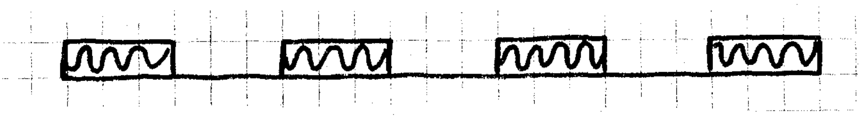
Illustration # 2 shows the various techniques used in studios and control rooms. Any combination of these may be used, as well.



2a. Splays



2b. Polycylinders

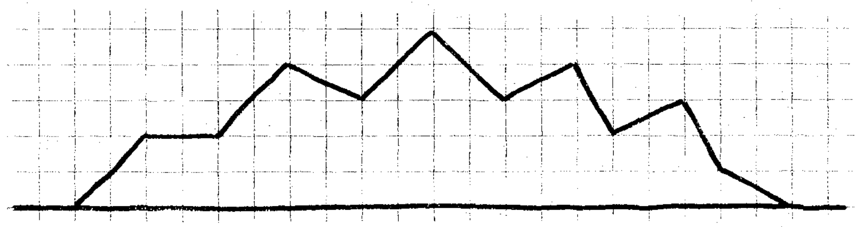


2c. Mixed Randomly Placed Absorptive & Reflective Surfaces

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

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

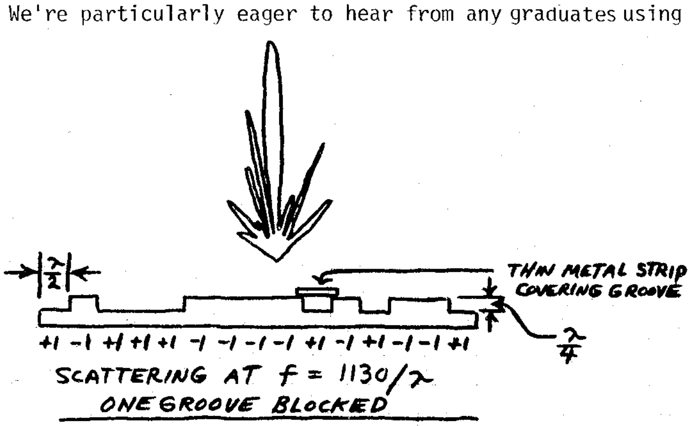
Four or five different panels designed for center frequencies, say of 500, 1000, 2000, 4000, and 8000 Hz should suffice for most spaces.

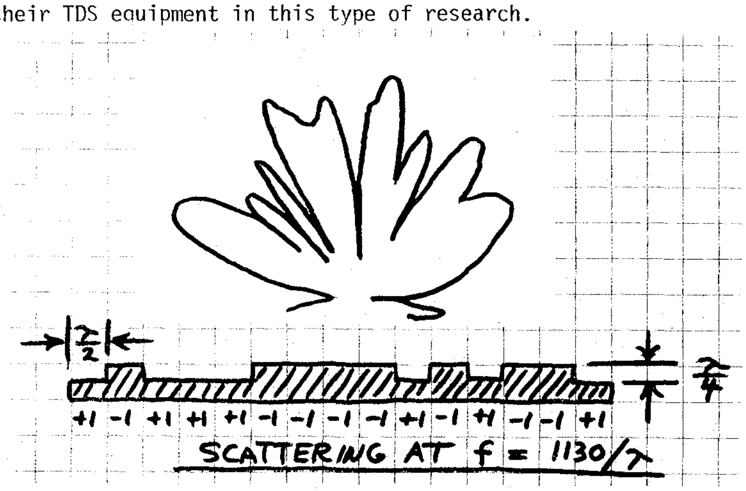


2d. Polygonal

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

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

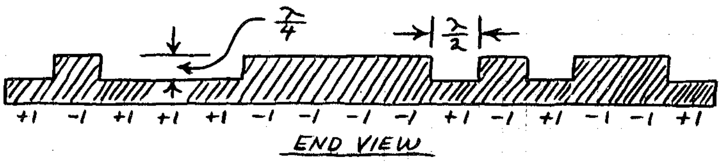


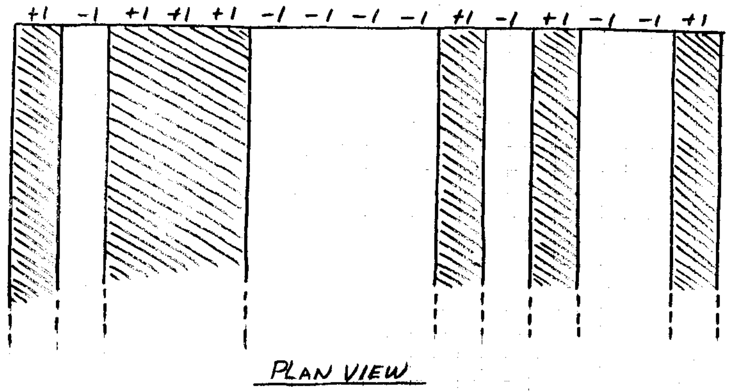


3a & b. Effect of blocking one groove in panel

Diffusion, continued

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





λ/4 = ONE QUARTER WAVELENGTH OF FREQ f

λ/2 = ONE HALF " " " " "

LENZ'S LAW AND MOTIONAL Z

In 1834 Heinrich Friedrick Lenz (1804-1865) deduced that when a coil is moved in a magnetic field "The induced current will appear in such a direction that it opposes the change that produced it."

In Beranek's Acoustics we find

$$Z_{et} = Z_e + \frac{Bl^2}{Z_m + Z_L}$$

Where: Z_{et} is the total electrical impedance of the unblocked diaphragm
Z_e is the electrical impedance with the mechanical motion blocked
Z_m is the mechanical impedance of the mechanical elements of the transducer
Z_L is the mechanical impedance of the acoustical load on the diaphragm
Bl is the product of the flux density of the magnetic field times the effective length of the wire cutting the lines of force perpendicularly

This equation reveals that an electro-magnetic transducer is an impedance inverter.

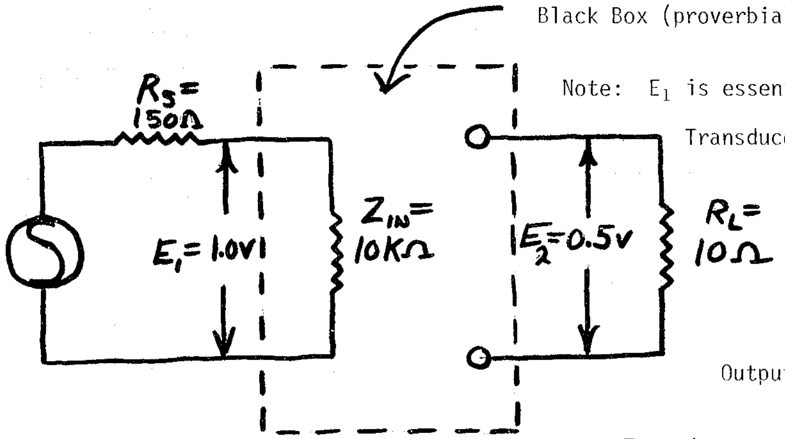
In the classes we measure Z_{et} and Z_B (blocked impedance) and as you will recall the resonance peak at low frequencies drops to Z_e when the woofer cone is held and not allowed to move.

When $Z_{et} = Z_e$ under such conditions we know that the motional impedance (Z_{mot}) is

$$Z_{mot} = \frac{BL^2}{Z_m + Z_L}$$

GAIN CALCULATIONS

Black Box (proverbial)



Note: E_1 is essentially an open circuit voltage

Transducer gain in dB = $10 \log \left(\frac{4(E_2)^2 R_S}{(E_1)^2 R_L}\right)$ =

$$10 \log \left(\frac{4(0.5)^2 150}{(1.0)^2 10}\right) = 11.76 \text{ dB}$$

Avail. Power = $10 \log \left(\frac{(1.0)^2}{150(.001)}\right)$ - 6.02 dB = 2.22 dBm

Output power = $10 \log \frac{(0.5)^2}{10(.001)}$ = 13.98 dBm

Transducer gain = 13.98 dBm - 2.22 dBm = 11.76 dB

SYNERGETIC AUDIO CONCEPTS

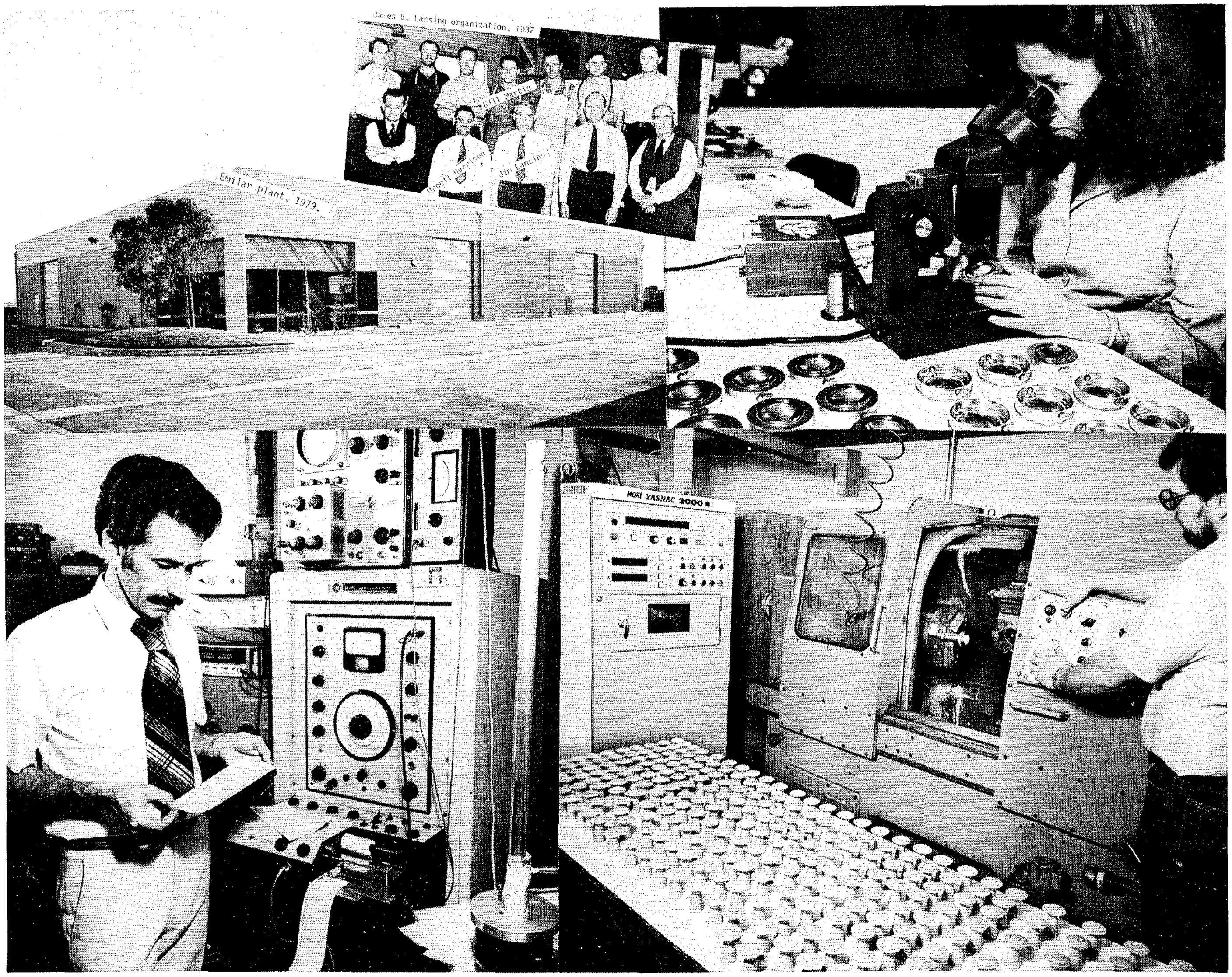
1979 FALL SEMINAR SCHEDULE

We have our Fall Schedule mostly established now. Because we can't go to every city that we would like to in a year, we rotate. Therefore, this year we will not be in Chicago, New York and D.C.

Minneapolis	-	September 10-12	Philadelphia	-	October 30-November 1
St. Louis	-	September 17-19	Nashville	-	November 6-8
Special	-	September 26-28	Orlando	-	November 14-16
Syracuse	-	October 9-11	San Antonio or Houston	-	December 5-7
Boston	-	October 17-19			

Our Los Angeles class at the Sheraton-Universal, North Hollywood has been filled since early April and we have scheduled another class in Los Angeles for June 20-22.

EMILAR EXPANDS AGAIN



I have been gathering material for an article on the history of the compression driver. John Humble, one of my associates at Altec for many years and now a successful independent manufacturer's representative, has arranged lunch at different times with several of the early workers in the field, Jim Martin, brother of James B. Lansing, being one of the very interesting meetings.

Among the material we have put together are pictures from the recently purchased Emilar facility. (Already a second facility of almost equal size has been leased nearby for expansion.) We thought we would share a few of the pictures here.

The picture of Jim Lansing was made in 1937. The pictures at Emilar were made in 1979. Both men associated with top talent. Both produced products distinguished for their precision and finish. Both knew how to set up precision hand-craft manufacturing plants.

The pictures tell a great deal -- especially if you have seen the facilities at some other major manufacturers of compression drivers.

HEYSER LA AES SECTION MEETING

Carolyn and I towed our truck and Airstream combination over 800 miles in 24 hours, returning from our Salt Lake City class, to attend Dick Heyser's paper at the March LA section of the AES. In this talk, Dick explained the basic philosophy behind the loudspeaker evaluations that he publishes in Audio Magazine.

One feature of a Heyser paper is the illustrations he uses to focus on each key point he has to present. We were treated to some exceptional three dimensional displays of the interaction of sound intensity, time, and timbre. The "warping" of timbre with intensity or of time with timbre were illustrated and the mechanisms causing such phenomenon were described.

As usual the immense void between Dives and Lazarus was apparent in that some of the loudspeaker manufacturers present didn't seem to recognize their own loudspeaker's flaws when illustrated. At one point Dick said to his listeners, "don't laugh, you may be looking at your own product".

While Dick labors to bring audio into the 20th century, just as physics was by Einstein and Bohr, et al, what perhaps is not realized is that the larger loudspeaker manufacturers are still back about the artisan stage exemplified by the Babylonian Culture of 1800 B.C.

Certainly the most satisfying aspect of our work is interfacing with the talent of our industry. Less that two weeks before the Heyser section meeting, we had Ed Long and Ron Wickersham into the San Francisco seminar to talk with the class. Being able to reflect on the information from their talks in such short time causes an explosion of new insights.

RAULAND ACQUIRES PHILIPS "NO HANDS" INTERCOM

Rauland has announced that they are now the *exclusive* distributor in the U.S. and Canada for the Philips M28 and M100 Duplex intercom systems. The significance of this move is that Philips, one of the world's larger corporations, has chosen a truly professional commercial sound manufacturer in the U.S. to represent them. That means that they recognize how well established in this market place Rauland is, and we're sure the future will see really fascinating developments from this initial relationship. Philips has done their homework and gone to the best financed, highest caliber distribution, most likely leader in the 1980s in commercial sound. Rauland has stated that they intend to seize the commerical sound market place in the 1980s. Philips has by this move, tacitly recognized the reality of that claim.

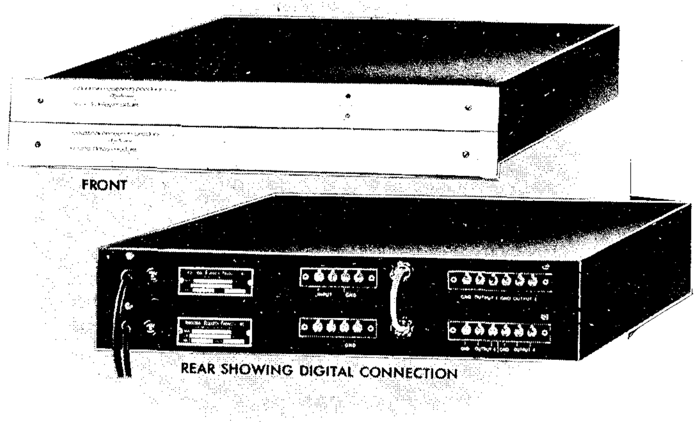
While we're delighted to see Rauland acquire this new line, it's the implied accolade that impresses us.

ANOTHER FIRST FROM INDUSTRIAL RESEARCH PRODUCTS INC.

Industrial Research Products Inc. of Elk Grove, IL produces digitial time delay devices that we preferred and used long before they became a Syn-Aud-Con sponsor. Our preference sprang from the technical leadership of men like Mahlon Burkhard and the resulting superior products such leadership inevitably produces.

Their latest innovation is another of those "Why didn't I think of that?" products. So obvious, yet not available in the market place prior to their introduction of the idea.

ANNOUNCING
DIGITAL TANDEM CONNECTION
FOR THE POPULAR DD-4012



A Basic Improvement in Digital Time Delay Devices

In a digital time delay device the signal goes through an analog-to-digital converter, then a series of shift registers, followed by a digital-to-analog converter. These converters are expensive. When two complete delays are added together there are more converters than are really required. With the new "digital tandem connection" for their very popular DD-4012 series, Industrial Research Products has made it possible to add them up through *straight digital connection* with no signal degradation and goes around the additional conversion step.

That means that instead of costing $950 for an additional delay line, it costs $240 for 4 outputs and 160 ms. Such cleverness we expect from the people who gave us the dynamic Threshold sensing on their new Voice-matic Microphone Mixer.

When you consider the growing importance and application of straight "in line" arrays down the ceiling of an auditorium, then a supplier like IRPI becomes a sought-after business connection. You can't help asking yourself, "What's next from the talented engineering department at Industrial Research Products?".

ED LONG - RON WICKERSHAM

At our San Francisco class in March, Ed Long and RON WICKERSHAM again presented some of their unique know-how with regard to their experiences with TA^{tm} and PRP^{tm} (Pressure Recording Process).

When you consider that Ed Long successfully explores the wild world of time warp in the crossover region without having an ETC analyzer, you begin to appreciate the care with which he observes oscilloscope data.

During his discussion with the class, Ed told many of the fundamentals bearing on successful time alignment^{tm}. Some of these are:

1. The larger the loudspeaker (more mass) the later in time it is likely to be.
2. Time warp is aggravated near the crossover frequency
3. Physical alignment can be helpful but sometimes misalignment physically can be used to offset electrical misalignment in the crossover network.

Ed defined minimum phase response as "the input predicts the output" and gave as an example cone breakup which is not predicted by the input signal but certainly shows up at the output.

After listening to Ed discuss TA^{tm}, we were aware of how powerful a tool ETC would be for such work. The precise acoustic centers could be easily found, time warp observed directly, and both network and physical adjustments could be modified and observed in real time.

One procedure would be as shown in Figure #1 & 2.

Ed Long talking to the class

FIGURE # 2. A MISALIGNED TWO - WAY LOUDSPEAKER

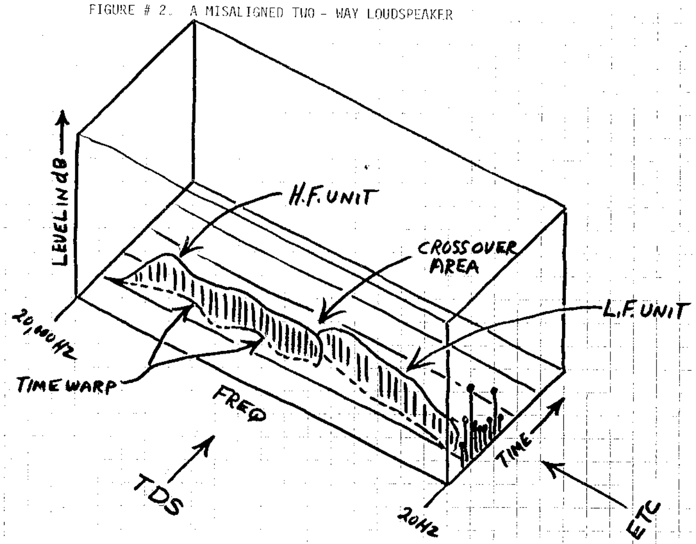
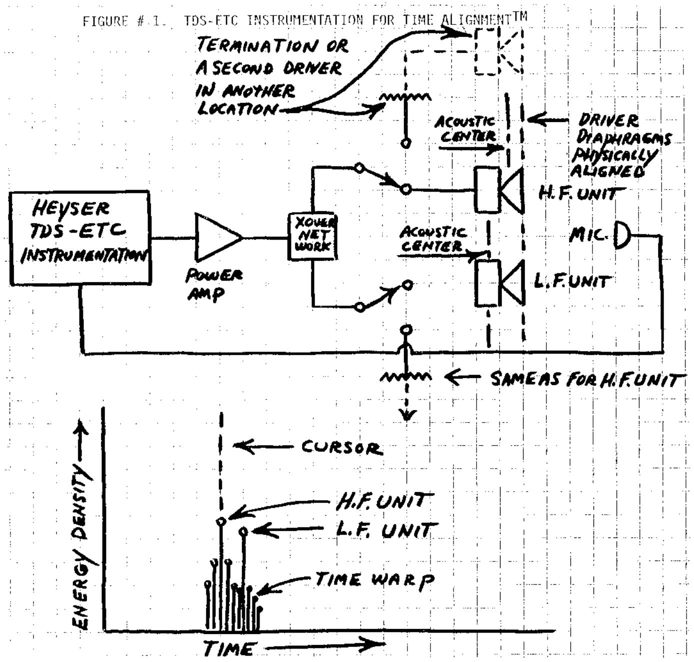


FIGURE # 1. TDS-ETC INSTRUMENTATION FOR TIME ALIGNMENT^{TM}



Ron Wickersham

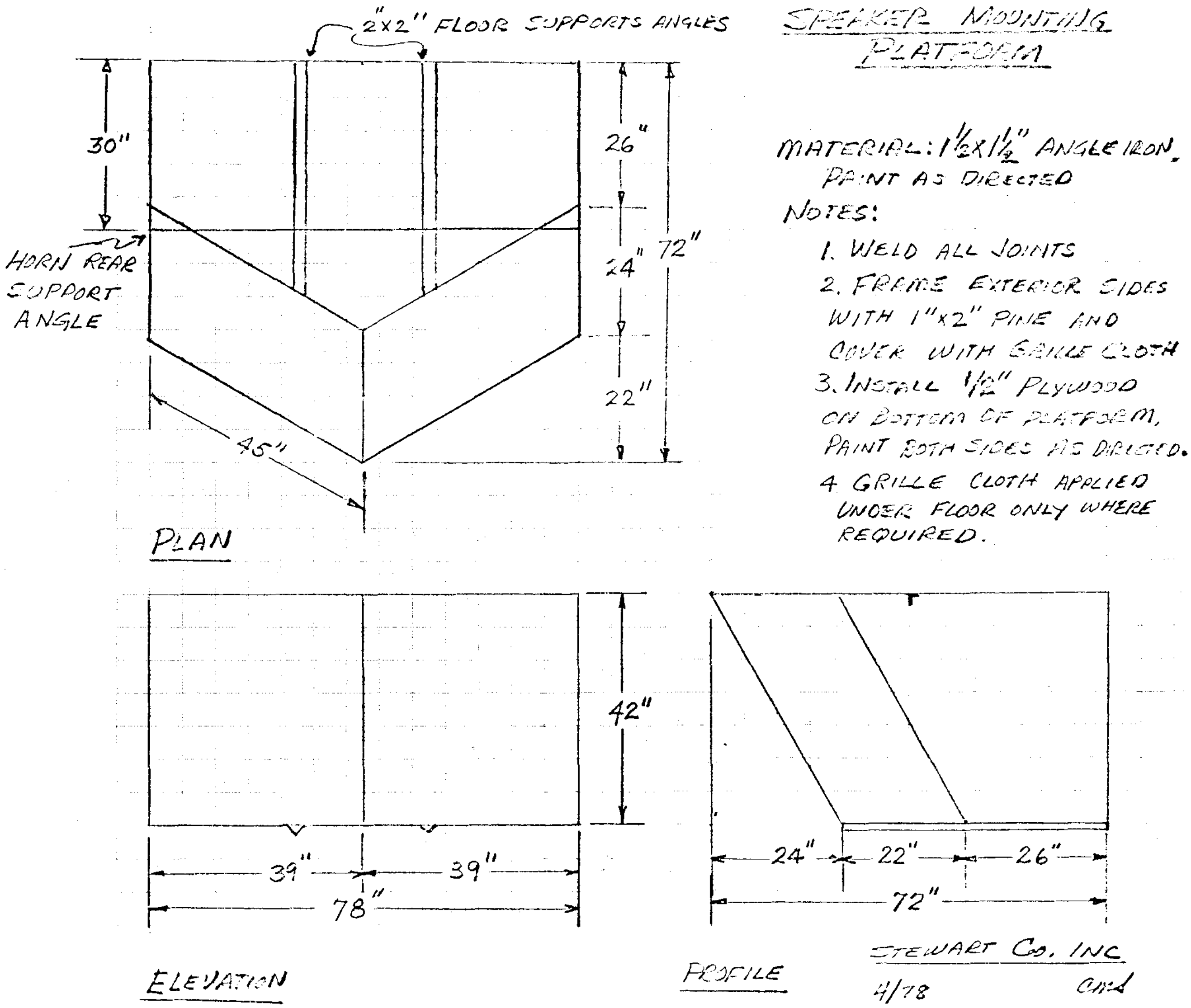
Switch on one driver at a time (preferably with a second real driver as the load for the network when the driver being aligned is turned off). Find out which driver is closest in time and set the cursor at that point. Then physically adjust drivers to best compromise and work between physical and electrical alignment by watching ETC analyzer.

Remember TA^{tm} is both trademarked and patented. Therefore, these ideas are put forward for experimentation only. We personally preferred to purchase a pair of Time Aligned^{tm} loudspeakers licensed by Ed Long for own use.

There's a lot more to a good loudspeaker system than just time alignment. Ed and Ron are very clever at everything from cone construction, network analysis, and enclosure design, to the psychoacoustic weighing of least worst distortion tradeoffs.

LOUDSPEAKER MOUNTING

CY STEWART of Stewart Company of Lansdowne, PA has given us permission to print a simple but effective speaker mounting platform drawing that he uses regularly in his work. We appreciate such sharing.



HEADROOM

ED LETHERT writes:

Here is something that I've been puzzling over for some time. A certain manufacturer makes an amplifier rated at 75 watts per channel. This power rating is continuous sine wave and corresponds to 24.5 volts RMS into 8Ω. The current at 75 watts will be 3.06 amps RMS. The product of the 24.5V (RMS) and 3.06A (RMS) is equal to 75 watts.

Now, at 24.5V (RMS) sine wave has a peak value of 34.6 Volts. Similarly 3.06 A (RMS) is a peak value of 4.33 amps. Multiplying yields a peak power of 150 watts. If we connect an 8Ω driver rated at 15 watts continuous and 150 watts peak to the amplifier output, it would seem that we could operate the system with as much as 15 watts of average program signal and still maintain 10 dB of head room. Conversely we could operate the system at the normally selected maximum of 7.5 watts program and have 13 dB of headroom. What do you think?

We believe that the confusion arises from the terminology "headroom". Tech Topic Vol 4 # 1, page 4 - Hans Schmid defines "meter lag" and "head room". The 10 dB "meter lag" we adjust for ignores "peak" values and is 10 dB below fully rated average power. "Headroom" is how much margin you have before the amplifier actually clips, and given short enough transients, it would be 3 dB for the system Ed describes.

We'll be pleased to hear from any graduates with other thoughts on the subject.

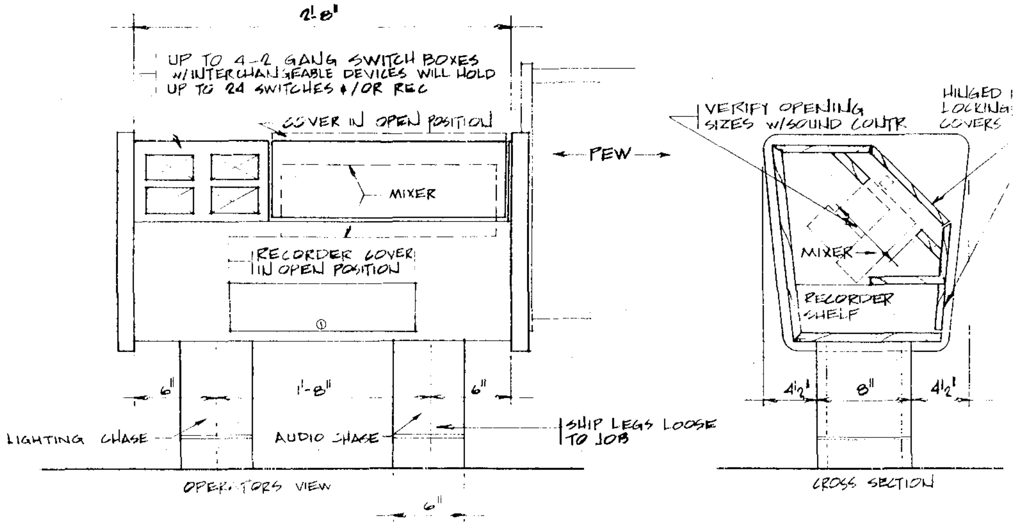
VERY QUOTABLE

JOE PAVONE sent in his check for a TDS license following the 1978 New York class, with the following words, which we like very much: "TDS will be a very useful tool in everyone's tool box. *My ears work together with my eyes when I enter a new control room now.*"

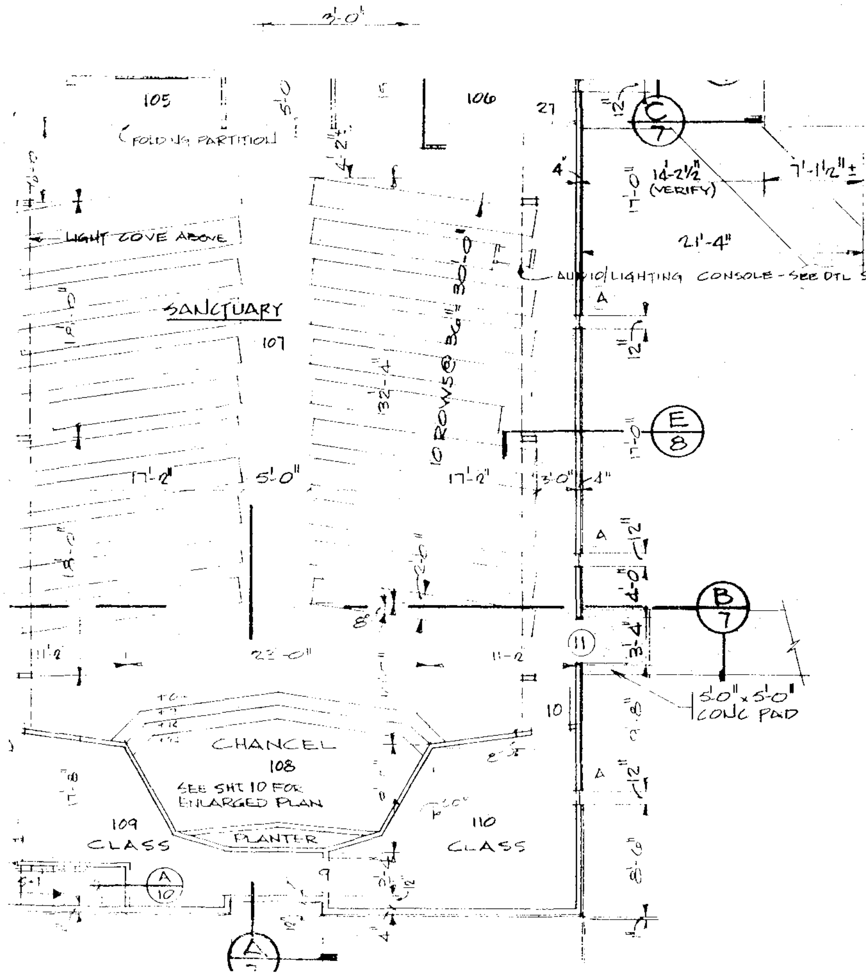
PLACEMENT OF SOUND SYSTEM CONTROLS IN SANCTUARY

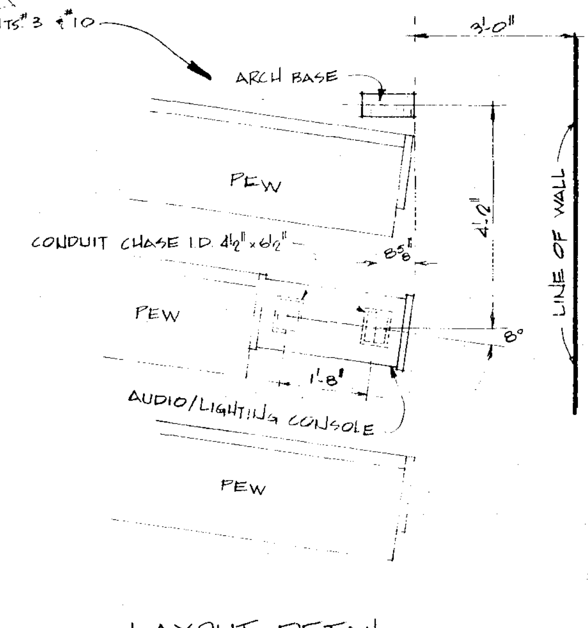
LES JORDON, Communications Manager at Goodman Church Builders in Joplin, MO, is in the enviable position of being able to instruct the pew builder (pews on their jobs are *normally* custom fabricated) as to where the sound control equipment is to go *in the congregation area* and have them custom build the housing for the controls as an integral part of the pews.

Les has been kind enough to share a typical set of drawings with Syn-Aud-Con and we are indebted to him for them. We hope that when you are dealing with churches you will go and do likewise.



AUDIO/LIGHTING CONSOLE
SCALE: 1½" = 1'-0"





LAYOUT DETAIL
AUDIO/LIGHTING CONSOLE
SCALE ½" = 1'-0"

JOHN MONROE ODUM is an unusual man with highly developed physical coordination, coupled with an adventurous mental attitude, backed by a generous intellectual capacity, a skilled hand-gunner, a mean man at a set of drums, and one of the advanced cases of computerholism.

COSMOLOGICAL DISCOTHEQUES

The discotheque, as has presently evolved, is a sociological theme that mankind has continually repeated throughout his entire existence on this planet -- from the most primitive forms of man to the present day being. In studying the ancient customs and rituals, you will find a recurring musical motif -- a strong pulsating beat. A beat similar in rate to the heartbeat of man -- a rate closely aligned to man's own rhythmic intercourse with the universe about him. Whether man understood the "captive" effect of the strong pulsating rhythm, or simply, intuitively guessed the effect is not known in the case of early man. But, nonetheless, the effect has been used most efficiently by both primitive and modern leaders to incite the passions of man for patriotic allegiance, for total loyalty and lust in battle, and to generally cement public relations for the purpose of consolidating "power"!

These techniques implemented in battle ceremonies, oft scheduled feasts, gatherings, church socials and celebrations, have been in mankind's history since long before the bow and arrow was a secret weapon -- feared by all as the prelude to the end of it all. Man instinctively understood the powerful "touchstone" that existed in his fellow man and was quite successful in implementing this knowledge in swaying and shaping the attitudes and opinions of others towards some decided goal. Whether or not the author of an event was striving for a political binding, the spoils of war, or simply a composer writing music to move his fellow man, you will find that a strong rhythmic feel is always present in a composition, whether it be the classical music of a Beethoven or the disco music of Gene Page!!

Understanding this "touchstone" principle can give a powerful tool to those who will use it. A tool that can influence people subconsciously in many ways, from implanting a friendly disposition towards a political candidate, or swaying public opinion towards an issue by a politician, to simply winning customers that will support your discotheque because the proper implementation of the sound system has further enhanced the "captive" effect of the "touchstone" principle.

Now, if this sounds a little farfetched, then examine your history books from past to present. You will discover that mankind has used this "touchstone" principle far more than you might imagine. This pulsating pattern occupies most all civilizations in the form of chants and mantras that actually induce a state of self hypnosis allowing gurus and even the simple folk to perform easy as well as incredible feats. These have ranged from ignoring hunger because of famine, to actually causing physiological changes in the body such as reduced oxygen consumption or slowing of the heart to an almost undetectable rate. This in conjunction with some rhythmic pulse to help induce the proper mental state for man to be able to accomplish these near miracles of mind and body control.

This "touchstone" chanting can be found in every land to a certain degree: Africa, China, South America, the Catholic Gregorian chants, the American Indian, Roman slave oarsman, any slaves, war chants, peace rally chants, marches -- in every walk of life you will find such examples. The more successful countries have always paid special attention to the music used in public ceremonies. You certainly won't get a man off to war with a weak tune -- what's needed is a strong rhythmic march to send him fervently on his way! All of our important events are heralded by strongly rhythmic music -- it's a fact of life - one that cannot be ignored - and one that can afford you great opportunity when coupled with present technology in sound reproduction.

So, now that we have accepted this natural "touchstone" principle, what can we in the disco business do to help insure that we capture our customer's loyalty on as many levels as possible, from service to the customer, to satisfying the customer's sensory inputs? Many things can be done involving the right mix of colors, size and shape of room, and the proper implementation of lighting. But let's look at the sound system and see what we need to do in general terms, to help create a successful "disco sound" that will utilize the "touchstone" principle to the max!

Studies have shown that the more successful disco systems are able to reach down to 30Hz and below with usable energy. It appears that the thigh area likes a low frequency massage with 14 Hz and 40 Hz being two critical reinforcement frequencies. Probably the most successful way of insuring massive amounts of low frequency without boosting the mid-range frequencies to the painful point, is to utilize sub-woofer systems directly coupled to the floor. This will afford you huge amounts of low frequency power to the dancers and will reinforce the primordal feelings of the "touchstone" principles on the subconscious. This is very important! Man has instinctively groped for that low frequency pulse by turning up volumes more and more, trying to boost the ever important "intimate massage" frequencies. But in doing so has also boosted the mid-range frequencies to levels that cause discomfort and pain to the ear. You should realize just how important these pulses are to man, for many have suffered severe high and mid-range frequency losses in their hearing while subconsciously trying for that powerful low frequency pulse by turning up the volume on what usually is a left hemisphere designed sound system.

Technology is a wonderful tool and can be implemented in a way congruent with man's natural principles of right and wrong. Augmenting and enhancing a natural "touchstone" in man, of which there are many - mother and apple pie to name one, can only be a benefit to one and all, and should be utilized to the max by people in any form of communication, from simple speech reinforcement to what well may be a more complicated form of communication -- the incorporation of lights and music in a coherent manner -- the discotheque!!

REFERENCES:

1. Arthur Janov, PhD, "The Primal Scream", Putnam's Sons, New York
2. Vincent Bugliosi & Curt Gentry, "Helter Skelter", W. W. Norton & Co., New York
3. Harris & Crede, "Shock and Vibration", McGraw-Hill, New York
4. Alex Rosner, "Comments from a Disco Expert", Syn-Aud-Con Newsletter, Tustin, CA.

ANAHEIM CLASS 1979

RICHARD LONG - "BEST DISCO SOUND DESIGNER" AWARD

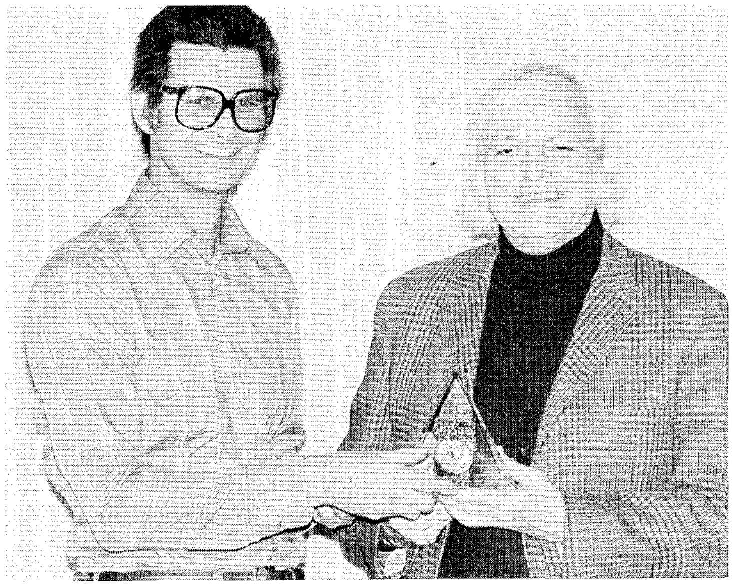
When we pass out the Syn-Aud-Con certificate at the end of the class, I always say, "If you do a good job, say 'Of course, I'm a Syn-Aud-Con graduate', and if you have a failure shove it into the right hand desk drawer." I hope Dick Long has all three of his Syn-Aud-Con certificates hanging on his wall along with his awards for "Best Disco Sound Designer" and "Best Disco Sound Installer" from Billboard Magazine's Disco Forum V.

Dick is truly a pioneer in disco work. He attended our first New York class in 1973 and his company name was Disco-Sound. Dick doesn't "fly by the seat of his pants". He is a professional with full instrumentation, including the Acoustilog reverberation timer and the HP 97 - with many programs.

The award for "Best Disco Sound System" was presented to Larry Levan, deejay of the Paradise Garage, which is also one of Dick's top sound installations. He can also take credit for Studio 54, Regines, New York-New York, Emerald City and Annabel's.

Some months ago, Business Week had an article "The Feverish Hustle for Big Disco Profits". The article mentioned Studio 54, the best known disco among 10,000 in the U.S., as having a full return on its investment (over $500,000 but under $1,000,000) in less than 6 months.

Dick's sound system is an important part of that success.



Richard Long of Richard Long Associates on the left receiving the awards from Billboard Magazine.

NEW PRODUCTS FROM SUNN MUSICAL, INC.

One of our earliest sponsors, Sunn is well known in the musical instrument marketplace but has never entered the commercial sound field.

Our contact with Sunn has been very beneficial to Syn-Aud-Con due to the very high caliber of personnel at Sunn both in engineering and marketing who communicate worthwhile new ideas every time we interface with them. The engineering at Sunn is young, talented, and also has been there long enough to be well proven.

They now tell us that they are readying a whole series of new products aimed at the pro-audio market. The new products will be introduced in June. They include:

1. Two new bass horns -- and Sunn has excellent know-how in this area.
2. Two new high frequency radial horns made with a new material that has never been used in horn construction, new injection foam-type material. (*Sunn has TDS* to check their design and construction.)
3. New 15" and 18" woofers
4. Two new power amplifiers using switching power supplies
5. A *digital* electronic crossover
6. A 1/3-octave graphic equalizer

Sunn is blessed with excellent management: Larry Lynn, President; Bob Yaruss, Sales and Marketing; and Rod Goldhammer, Engineering. The advent of innovative products in these new areas makes them worthy of investigation not only as a source of new products but a possible prime supplier as both of you grow in the industry. If you are not now familiar with Sunn we suggest you get acquainted. One thing we can be very sure of -- you'll like the people you meet and you'll respect the products they offer.

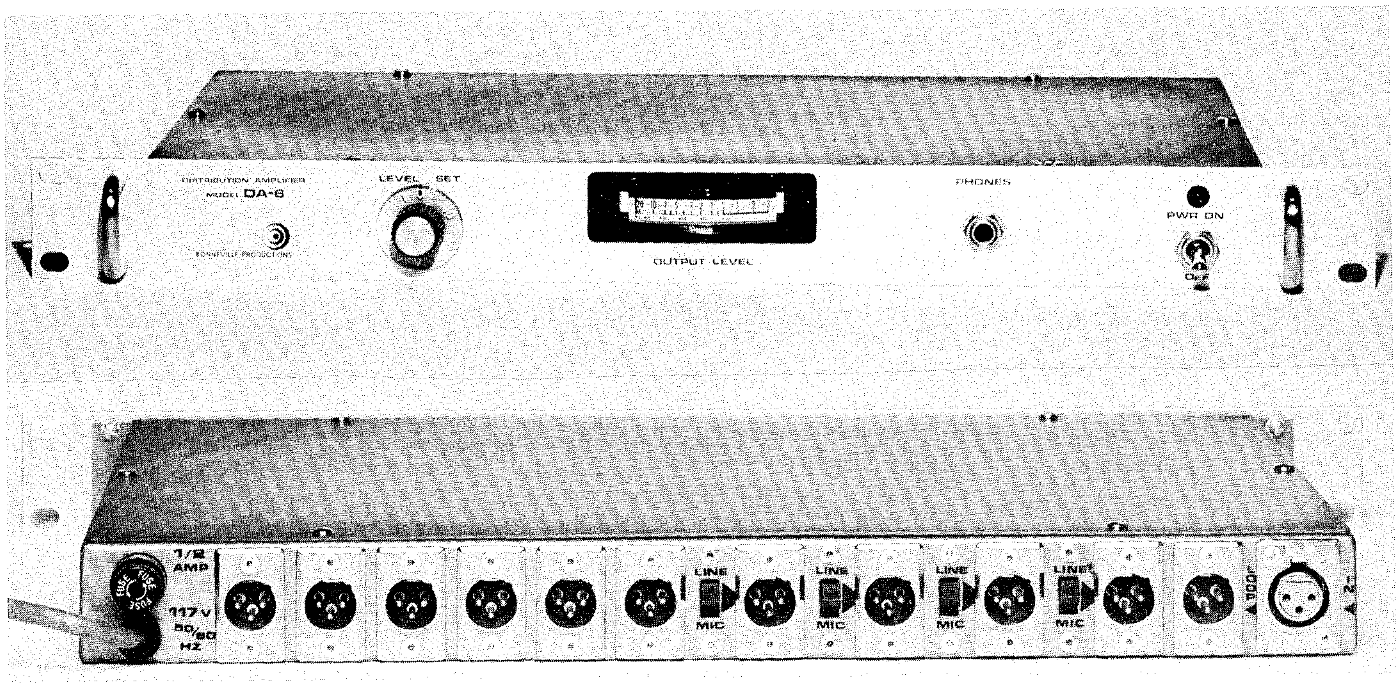
PB 50 WORD LIST

JERRY FRIEDLAND of Ace Sound Rental Company in New York sent us the PB 50 Word List, which he says is presently used by ear, nose and throat doctors when they administer hearing tests. Obviously it can be used for checking sound systems for articulation.

1. feat	11. use (yews)	21. rag	31. slip	41. death
2. rise	12. folk	22. there	32. crash	42. nest
3. hive	13. is	23. wheat	33. not	43. then
4. rat	14. are	24. cane	34. heap	44. fern
5. cleanse	15. smile	25. ride	35. no	45. strife
6. fuss	16. dish	26. bad	36. plush	46. fraud
7. nook	17. bask	27. pile	37. toe	47. hid
8. such	18. creed	28. mange	38. bar	48. ford
9. deed	19. rub	20. hunt	39. pan	49. grove
10. pants	20. box	30. clove	40. end	50. dike

DISTRIBUTION AMPLIFIER FROM BONNEVILLE PRODUCTIONS

MIKE COLLETT of Bonneville Productions in Salt Lake City showed us their new line of distribution amplifiers and line/isolation amplifier. Since good distribution amplifiers are not easy to find we are pleased to have a Syn-Aud-Con grad's product to suggest you check out if you have a need.



Mike gave us the following information for the Newsletter: An audio distribution amplifier with XLR connectors is now available "off the shelf" from Bonneville Productions in Salt Lake City.

The unit is particularly suited to press-conference use to provide multiple line or microphone outputs to reporters and sound cameramen. It is also useful for multiple feeds in portable sound reinforcement systems. It finds use in mobile recording on location, interfacing large audio systems at conventions and media-covered events, and isolating feeds in applications where A-C ground currents might present problems.

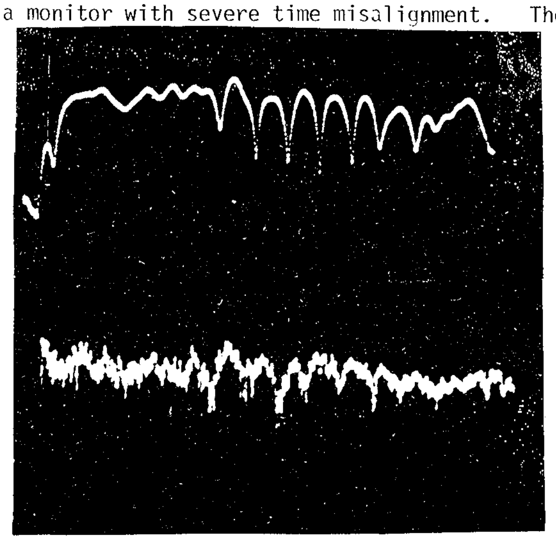
The DA-10 features up to 30 dB of gain, low output noise (-80 dBm), high headroom (+24 dBm, loaded), low distortion (less than 0.5% at +24 dBm out) and flat response (±0.5 dB 20-20KHz).

The amplifier is self-contained with its own power supply, features transformer coupling on all inputs and outputs, and provides a VU-meter and a headphone jack for signal monitoring. Ten outputs are provided: six are line output and four are selectable mic. or line level.

The DA-10 is available from: Bonneville Productions, 130 Social Hall Ave., Salt Lake City, Utah 84111. Telephone, (801) 237-2400.

M_a FACTOR OF 65

There is currently in use in many recording studio control rooms a monitor with severe time misalignment. The top curve in the photograph is the TDS response of this monitor from 0 to 20,000 Hz and the six major anomolies are 1,000 Hz wide and start at roughly 8KHz and end at approximately 13KHz.



At 8KHz a 1/3-octave band is 1840 Hz wide, so this anomoly is 1/6th octave at that frequency.

These anomolies are clearly audible in a free field and in this studio that was using this particular monitor had a total sound that looked like bottom curve of the photograph. Note that the anomolies are as evident in the total field as on the TDS. This is because the rear wall in the control room had an M_a factor of close to 65.

DOES A HIGH Q DIGITAL FILTER RING ?

Lee Irvine and JIM FULLMER (Jim has been through many Syn-Aud-Con classes), Acoustical Consultants, Inc., in Salt Lake City, were our hosts at a dinner with Dr. and Mrs. Stockham of digital recording fame. The Irvines and Fullmers were ideal hosts and had engaged a private room at a supper club that allowed spirited conversations with fascinating people.

Among the many questions we had been saving for such an occasion was the one we asked Dr. Stockham, "Can a digital filter be designed to violate analog filter "Q" restraints (such as ringing)?".

The answer is, "No, it can't. A high Q digital filter will 'ring' for the same length of time as an analog filter will under shock excitation. It's the mathematics that govern and not the circuit type."

Dr. Stockham was a wellspring of interesting viewpoints. His description of the decibel was graphic. He held his hands 10" apart. Then he brought them to 9" apart. "That's 1 dB," he declared, "anybody can *see* that."

Dr. Stockham had attended MIT in Professor Norbert Weiner's day and he added to our collection of Weiner stories. It seems that Weiner had moved to a new residence. At the end of his first day at work he had quite forgotten that he had moved and drove home to the old house, now empty. Seeing some children out in front playing he went up to them and asked, "Children, can you tell me where Professor Wiener has moved to?" One of the youngsters replied, "Sure, Dad. We live around the corner now."

Dr. Stockham also told of having lunch with Professor Weiner the day that Joe McCarthy died. He said that Professor Weiner shook his head sadly and said, "I wish I believed in Hell today."

THE SHURE SM82 IS A CARDIOID

It's painful to make a mistake in the Newsletter, but it has its reward. We find out that quite a few people are reading the Newsletter. Back in Volume 6 # 1 we said that the Shure SM 81 was "to our knowledge the first condenser microphone that Shure has ever offered". We corrected that mistake in Volume 6 # 2 by saying that Shure's first condenser microphone was the Model 400 introduced in 1935, and that Shure's first modern condenser microphone is the SM-82, which was introduced in 1974. Then I went on to call the SM-82 an omnidirectional. That brought more letters. The SM-82 is not an omnidirectional, but cardioid. PAUL BUGIELSKI who heads up Professional Sound Products at Shure Brothers said he was sorry to make our face red again, then he added some new "laws" to lighten the day:

Harris's Law--Any philosophy that can be in "in a nutshell" belongs there.

Perversity of Production Precept--If it works well, they'll stop making it.

Shaw's Principle--Build a system that even a fool can use, and only a fool would want to use it.

HEAVY READERS ARE HEAVY HITTERS

In the September 1978 IEEE Spectrum a report on a survey of 3,000 RCA engineers revealed some important habits that correlated directly with success in technological endeavors.

The group identified as "high achievers":

1. Accepted the premise that keeping current in their field, not just in their job, was important
2. Compared their information gathering habits with those exhibited by other identifiable high achievers and took specific action to emulate their peers
3. Read more technical, business and professional journals than low achievers
4. Used libraries and collected books
5. Sought out experts in other areas than their own
6. Participated in technical symposia, workshops, and meetings.

Many high achievers felt they could use more math. All felt they could use more time for study and reading. The statement is made in the report that "Because high achievers consistently value and use more information and because reading is an important method it's not surprising to find differences in time spent reading. (as compared to low achievers)

As a Syn-Aud-Con graduate you'll find you fit the high achievers profile remarkably well. You *are* working on the habits that lead to excellence of performance. Reading this Newsletter is proof of your extracurricular reading habits. You spent time and money to attend a Syn-Aud-Con class. You're interested enough in your field of audio to look far beyond just the daily demands of your job. Congratulations.

COMPUTER-ANTIMATED FILMS ON ACOUSTICS

Penn State's Graduate Program in Acoustics has prepared 4 computer-animated motion pictures on acoustics. All films are 16 mm, color with optical or magnetic sound. The films are available on a purchase-only basis and must be prepaid.

Simple Harmonic Motion (16-1/2 min.)	$200
Sound Fields in Rectangular Enclosures (14 min)	180
A Brief Colloquy on Acoustic Diffraction (11 min)	160
Process of Holography (10 min)	150

Make checks payable to The Pennsylvania State University.

BOOKS OF INTEREST

PIERCING THE REICH by Joseph E. Persico, published by the Viking Press, $14.95.
"Those who fail to learn from history are doomed to repeat it" is demonstrated century after century. The past few years, thanks to the British 30 year law clearing a whole series of secrets from World War II, has resulted in a flood of books written, in many cases, by the participants themselves revealing for the first time what really happened. I have enjoyed collecting such books and taken together they teach the reader a great deal about today's probabilities, as well.

One of the most recent of these books is *PIERCING THE REICH*. The book contains the authentic story of the OSS and their penetration of Germany itself during the later stages of World War II. It's not a book for the squeamish or for those who feel that meeting fire with flamethrower is not gentlemanly. This book is about the hard, dirty, very dangerous, and almost completely thankless task of saving the civilized world done by men risking everything alone in a vicious police state where death, when it came, was unbelievably barbarous.

There is an entire chapter on Syn-Aud-Con graduate, STEPHEN H. SIMPSON, JR. of San Antonio, TX relating how he developed, established, and demonstrated a radio system for use by agents in Germany that was not detectable by conventional radio direction finders.

Steve risked his life to prove the system by personally flying the missions into Germany in old beat-up British De Havilland Mosquito bombers through storms of flack, terrible weather, and the very real danger of being shot down by our own fighters (Steve removed even the IFF - Identification friend or foe radio equipment - in order to save precious weight and add to the plane's time over the rendezvous point). Steve's radio for the agents sent their signal straight up with nothing to the sides, thereby making them undetectable on the ground but easily read by an orbiting airplane at 30,000 ft.

There may be phases of peacetime intelligence work that need some review at times but there very few politicians in our history worthy of the type of men described in this book.

The twelve books I feel lead the reader to a maximum of facts and a minimum of fantasy are:

1. *Piercing the Reich*, Joseph E. Persico. The Viking Press
2. *A Man Called Intrepid*, William Stevenson. Harcourt, Brace, Javanovich
3. *Most Secret War* (British edition); *Wizard War* (American edition), R. V. Jones. Hamish Hamilton
4. *The Game of the Foxes*, Ladislas Farago. David McKay Co., Inc.
5. *Aftermath*, Ladislas Farago. Simon & Schuster
6. *The Code Breakers*, David Kahn. MacMillan
7. *The Service - The Memoirs of General Reinhard Gehlen*, translated by David Irving. World Publishing
8. *The Man Who Broke Purple*, Ronald Clark. Little Brown and Co.
9. *The Ultra Secret*, F. W. Winterbotham. Harper and Row
10. *The Nazi Connection*, F. W. Winterbotham. Harper and Row
11. *The Double Cross System*, J. C. Masterman. Yale University Press
12. *The Craft of Intelligence*, Allen Dulles. Signet (paperback)

These volumes can be quite an eye opener for those who think there are "rules in a knife fight".

******

THE ORIGIN OF CONSCIOUSNESS IN THE BREAKDOWN OF THE BICAMERAL MIND by Julian Jaynes. University of Toronto Press.
NICK METAL of Vancouver, who is one of the most erudite Syn-Aud-Con graduates, sent us a copy of *The Origin of Consciousness....*

Jaynes' thesis is a startling one - that catastrophe and cataclysm forced mankind to *learn* consciousness, and that happened only 3,000 years ago. Based on recent laboratory studies of the brain and a close reading of the archaeological evidence, psychologist Julian Jaynes shows us how ancient peoples from Mesopotamia to Peru could not "think" as we do today, and were therefore not conscious.

Jaynes feels that because the ancients were unable to introspect, they experienced auditory hallucinations -- voices of gods, actually heard as in the Old Testament or the Illiad -- which comes from the brain's right hemisphere, told a person what to do in circumstances of novelty or stress. This ancient mentality is called the bicameral mind.

Julian Jaynes examines three forms of human awareness -- the bicameral or god-run man; the modern or problem-solving man; and contemporary forms of throw-backs to bicamerality: hypnotism, schizophrenia poetic and religious frenzy, among other phenomena.

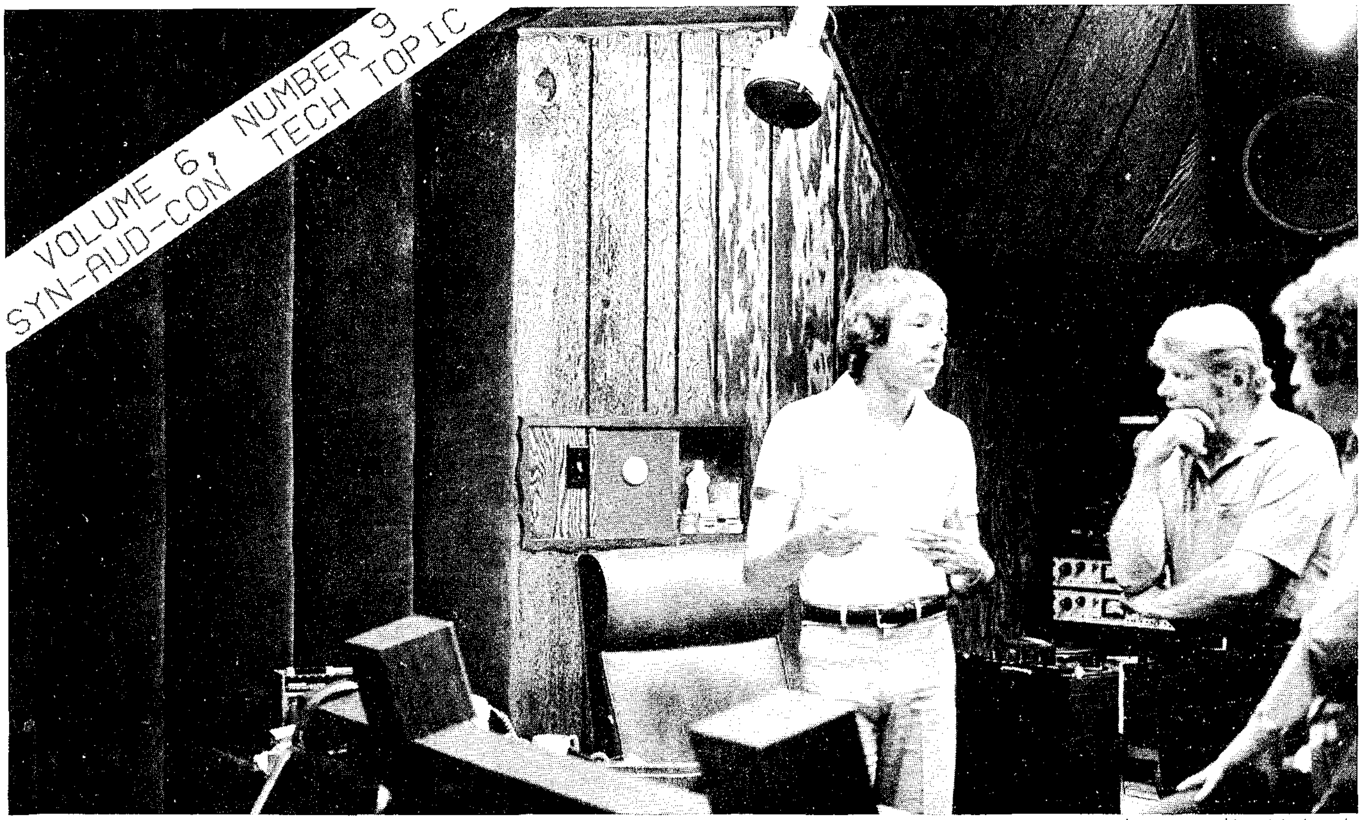
Anyone who has ever attained a set of skills, tuned to a hazardous activity, knows that at moments of extreme stress the unconscious "reflex" level takes over as when one gains control of an "out of control" race car or motorcycle. It is not uncommon to "see things" and "hear things" from deep out of the brain's memory bank.

In the words of one reviewer, it is "a humbling text, the kind that reminds most of us who make our living from thinking, how much thinking there is left to do."

Perhaps the most intriguing of all is the question, "Which hemisphere is nearest reality?"

*****

One of Tabs most useful books is *THE MASTER HANDBOOK OF ELECTRICAL WIRING* by Art Margolis. $6.95.
This 400 page book is basic-basic. It's for the reader who has looked into Crofts, *AMERICAN ELECTRONICS HANDBOOK* by Clifford C. Carr (former editor Terrell Croft) with a gasp and said, I need something more simple. This is it! You'll end up back at Crofts eventually but you'll probably have done a lot of very practical wiring in the meantime with the help of this book.



Chip Davis showing his **LEDE** control room to Bill Putnam.

Photo not used in original article

(LEDE) LIVE END - DEAD END CONTROL ROOM ACOUSTICS . . . **(TDS)** TIME DELAY SPECTROMETRY . . . **(PZM)** PRESSURE ZONE MICROPHONES

by CHIPS DAVIS
Las Vegas Recording

My original reason for attending Don Davis' Syn-Aud-Con class in February, 1978 was to learn about TDS (or time delay spectrometry). Little did I know at the time that I would take Don's findings on the early order reflections in hard-front control rooms and expand on the theory and make Las Vegas Recording, Inc., the first Live End - Dead End (LEDE) control room.

TDS

TDS was invented by Richard C. Heyser (U.S. Patent #3466652) and requires a special license to practice.* Heyser conceived and patented this vast improvement on pulse testing which had been in extensive use for 40 years, and called it time delay spectrometry. Briefly described, the receiver or tracking filters are delayed in time and do not start the receiver sweep until the signal reaches the microphone. This time delay sweep can then see the direct wave without having any interfering room reflections. (60 dB of signal-to-reflection.) It can delay the receiver and open the receiver window for longer periods of time until the first reflection is shown on the screen of the analyzer. The frequency, the depth in dB, can be seen and the time delay can be calculated to determine the surface from which the reflection came. Tuning can continue out in time until there are no other reflections, or the window is so wide only the total sound of the room can be seen. TDS can also show the amount of

continued on page 42 . . .

* Those wishing a license to practice TDS should send a check for $100.00 made out to California Institute Research Foundation, but mailed to Syn-Aud-Con, and a check for $25.00 made out to Syn-Aud-Con, for a set of "how to" notes on TDS. Syn-Aud-Con, P.O. Box 1134, Tustin, California 92680.

by DON DAVIS
Synergetic Audio Concepts

There are a series of fundamental "first principles" that underlie the LEDE concept. When Chips Davis undertook the redesign of his control room at Las Vegas Recording, I had only discussed the first of these principles — that of creating narrower anomolies. What is of great interest to me is that Chips in working out each new detail of the concept had "felt" what we are now actually measuring and mathematically developing. It is our belief that artist-engineers like Chips, when exposed to acoustical training, fulfill the promise inherent in recording technology.

First Principles

What are the "First Principles" underlying the success of the "Live end - Dead end" (LEDE) technique of recording studio control room design? The answer to this query lies in understanding how sound spectra propagate, reflect, undergo absorption or transmission, and combine. This is why you **always** see Time Delay Spectrometry (TDS) involved wherever a LEDE control room is engineered.

Time Delay Spectrometry

TDS was developed (and patented) by Richard C. Heyser in 1968. TDS allows the user to observe the direct sound level as if in an anechoic room with a 60 dB reduction of the signal from any surface. Since conventional anechoic chambers of first quality can only provide a 20 dB reduction it can be seen that TDS is an extremely powerful analytical tool of unprecedented resolution.

In observing the direct sound level with the TDS analyzer it is easily seen that no "room modes" are involved, even though the loudspeaker is indeed indoors because there are,

continued on page 52 . . .

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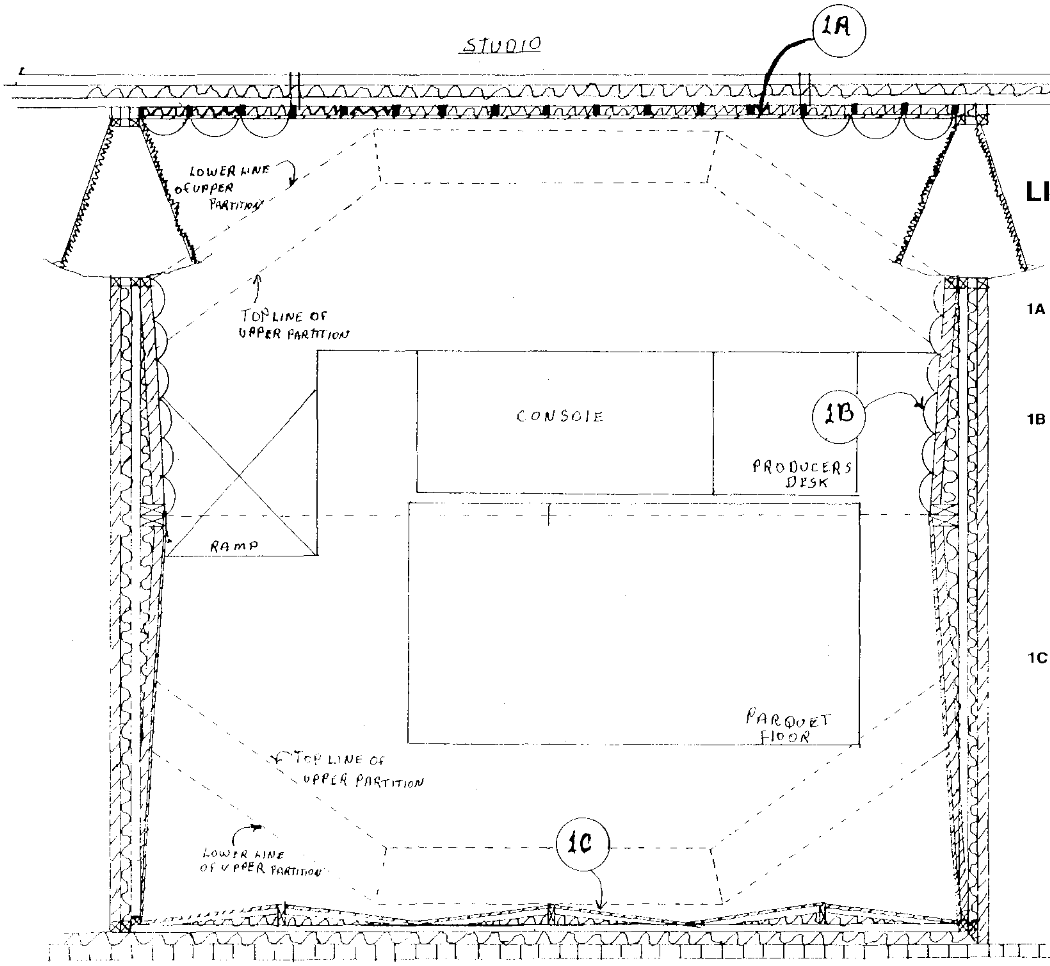


LIVE END - DEAD END (LEDE)

1A — DEAD LOWER WALL BELOW STUDIO WINDOW
Button board covered with fiberglass and covered with carpet
(Made to stand abuse.)

1B — CONSTRUCTION OF THE ABSORPTIVE END OF THE CONTROL ROOM
Standard studio wall construction will be followed. The absorptive battens will be constructed as follows: 1/4-inch plywood cut to width and length as determined by the decorative design. Cover plywood with R19 fiberglass. Cover fiberglass and board with heavy cloth. The batten will be nailed with paneling nails and glued with paneling adhesive. Fire retardant should be sprayed on all cloth coverings.

1C — THE LIVE END CONSTRUCTION
Standard studio wall constuction shall be used as with compliance to studio sound and deadening techniques. Live end decorative finishing materials shall be of real wood whether paneling or board planking — nailed and glued so as not to have any rattling or resonance effect

absorption various materials provide, the effectiveness of splays and not only acoustical problems but anechoic displays of speakers and microphones. An anechoic chamber only has about 20 dB of signal-to-reflection; TDS has 60 decibels of signal-to-reflection. Polar response of speakers and front-to-back ratios of microphones can thus be easily viewed with TDS. TDS is probably one of the most useful tools available to us today to find and analyze problems in our audio industry.

Live End - Dead End (LEDE) Control Rooms

Live end - dead end (LEDE) refers to the newly conceived acoustical design for control of rooms. It seems to me that this is the first real advance in recording studio control room acoustics in quite some time. To quote Don Davis, of Synergetic Audio Concepts, whose theoretical concepts of LEDE developed through his work with time delay spectrometry, those which compelled me to build the first live end - dead end control room: Don wrote, *"Pick up any book with pretensions of knowledge about recording studios and almost without exception the material on the internal acoustics exhibits an enormous void of accurate or useful information. Implied is that all you have to do is add absorption with the aid of some devil's apprentice with info from the dark domain and all is well."*

LEDE is basically the complete opposite of all other control rooms. That is, the rear of the room is hard and reflective while the front is as absorptive as possible.

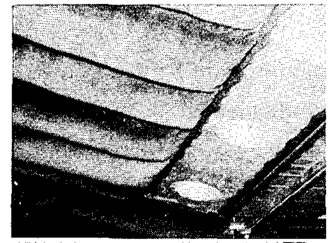
Let's start with the front of the control room, and explain the reasons behind the absorptive half. Davis found, through TDS, that mixing of early reflections from the hard ceilings and walls of conventional control rooms with the direct wave causes very deep anomolies in the order of 25 to 30 dB. (Anomolies are any deviation from the original response, therefore, distortion.) These anomolies are broadband and very deep when generated by very early reflections. They occur from the low mid to the uppermost frequencies beyond the audible range. The anomolies, from improper acoustical design, are caused by addition and cancellation of signals arriving at the mixing position out of phase, the phase depending on the time interval or the distance of the early order reflections.

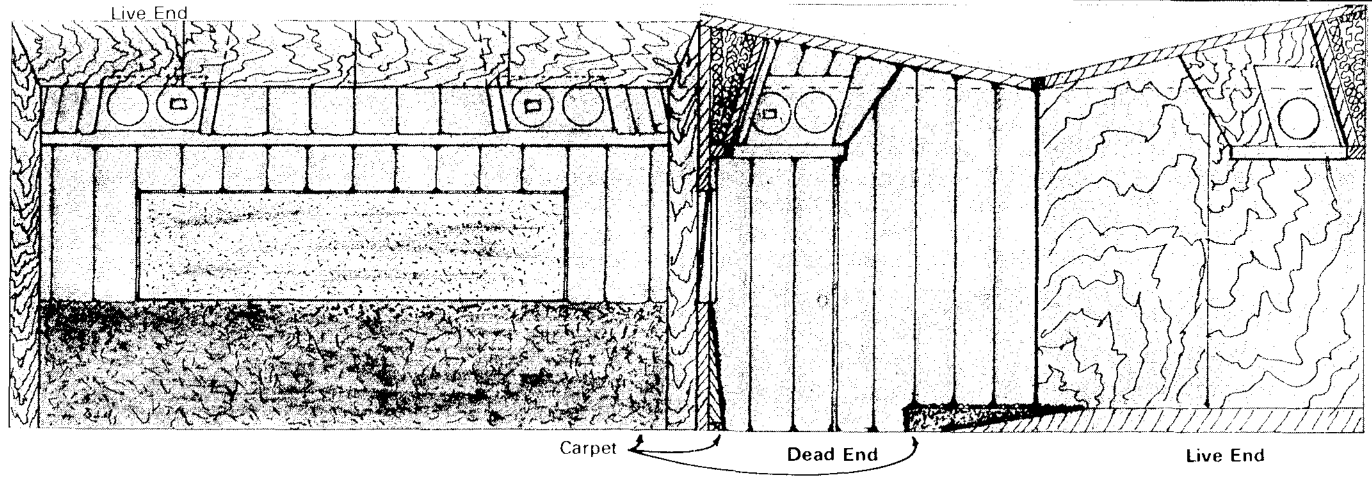
The acoustical anomolies and anomolies due to improper speaker design cannot be equalized into a smooth, flat reproduction spectrum. To equalize a control room under these conditions with the equalizing microphone at one position (in the mixing position), you could obtain a reasonably flat response. Move the microphone two inches and the curve becomes a gross, mis-adjusted, unequalized mess. Try this in your control room. Move the microphone in the area of the mixing position and watch the response curve change.

LEDE acoustical design minimizes this effect and helps keep a uniform frequency response in the mixing position. These anomolies are real and do exist in hard-front control rooms. We can see these effects and mathematically study their cause and effect with the aid of time delay spectrometry.

The Live End of LEDE

The live end of the control room is, I think, the most important part of the room. Davis gives a demonstration of the Haas effect in the Syn-Aud-Con class. It is a simple, but





very important fact of the LEDE control room. The Haas effect is the ability of the brain to discriminate against echoes and delays of sound that arrive approximately 10 to 20 msec after the original waves. The sound is still present but psychoacoustically does not exist so far as the listener is concerned. If the listener is 10 feet or less from a wall, the sound wave travels past him to the hard wall and back — a total of 20 feet — and he will not be aware of its origin. This is called the Haas effect. At greater distances the listener hears echoes or flutter. A hard-backed wall that is 10 feet or less away does not acoustically exist in our brains. The brain doesn't recognize or receive it. Again, this is the Haas effect. Therefore, we have, for the listener, eliminated the back wall, an infinite distance in space, psychoacoustically, and all we can hear is the front speakers.

Control rooms with very dead back walls compound other problems: room acoustics, speakers, and sloppy studio construction. Now that we have a disappearing back wall, we have to treat it acoustically, and this is where everything becomes like a game of acoustic pool at 1,130 ft. per second. We splay, angle, direct and bounce the sound that strikes the rear wall back to the mixing position. This stacking of the immense number of reflecting paths from the back wall is very precise and is figured extremely close as to time interval.

What we are trying to achieve is a very dense and diffuse total sound spectrum by combining the paths off the back wall into a series of controlled narrow band comb filters. Successfully done, the overall result is a very smooth total sound spectrum without any broadband anomolies. This procedure also masks console reflections, tape machines, people, etc., so that what is heard by the mixer is true, extremely accurate sound.

If the back wall is designed incorrectly, the possibility of having reflections arrive outside the 20 msec time interval would be disastrous. Inside the 20 msec range, an initial time delay gap of a much larger room is present at the mixer's position. You can turn and face the rear wall, cup your ears, and none of the sound from the monitor speakers ever seems to come from anywhere but the monitor

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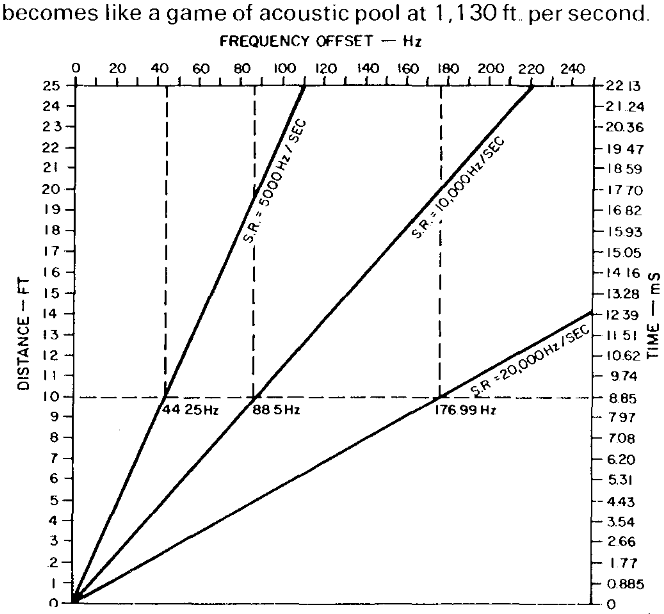
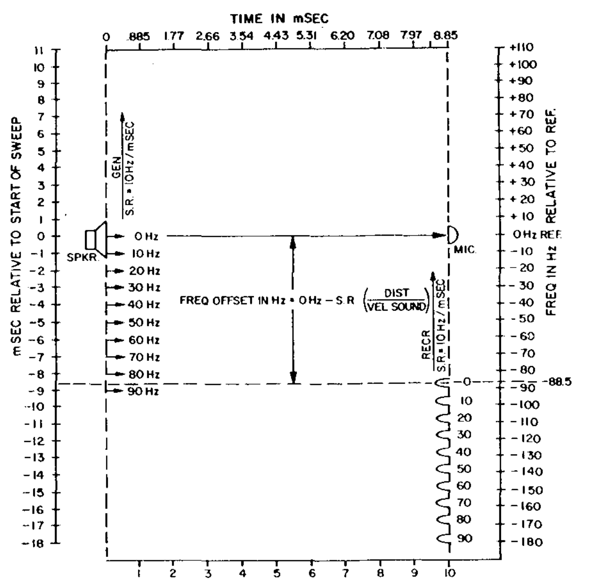


Figure 1. The Relation of Sweep Rate to Time.



Figures not used in original article.

Figure 2. The Relation of Generator Offset to Filter Sweep in a TDS Analyzer.

. . . continued from page 44:

speakers. It is totally undetectible in direction but audible in level. Careful diffusing of the rear wall and a very soft, nearly anechoic front wall are what makes an LEDE an incredible mixing environment. You have complete control of placement, depth and locality.

Time Alignment of the Monitors

LEDE and TDS are what we have put together so far. Now we will add time align™ (trademarked by E. M. Long Associates) monitor speakers invented by Ed Long and Ron Wickersham — the UREI 813s.

The 813s give a realism in the LEDE control room like we have never encountered. Before having installed the UREI 813s, we had a very popular, well known monitor. After looking at this speaker with TDS (See Figure #1) we decided

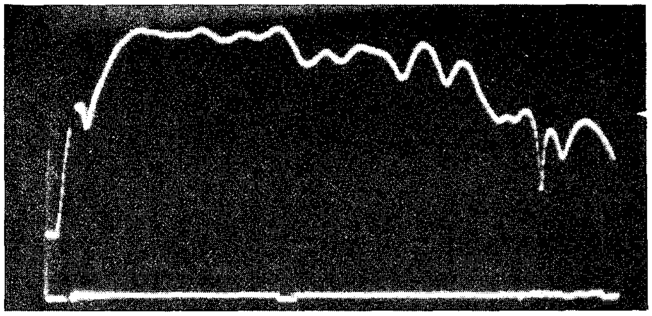


Figure I. Studio monitor which exhibits a series of severe acoustic anomolies. Measurements made with TDS.

to go to the 813s. Comparisons proved the 813s far superior due to the problems of mis-time alignment caused by mis-design of the crossovers. The problem cannot be easily eliminated acoustically or electrically. This distortion in the old monitor speaker could readily be heard by placing your hand over the tweeter and hearing the anomolies that we were seeing on the analyzer disappear. (See Figure #2) This

Figure 2. Studio monitor response when the H.F. unit is covered with a hand.

distortion is due to two elements emitting the same frequencies from different planes — the mid-range and the tweeter — with the crossover not cutting the mid-range off but allowing it to share a mutual portion of the spectrum with the high frequency. The effect is readily detectible and the ears can hear the distortion on cymbals and mid-range upper spectrum. You can hear the crackling of the anomolies as they phase in and out, causing the distortion feeling in your ears after you know it is there and focuses your attention on that particular section of the spectrum.

PZM™ — Pressure Zone Microphones

Having eliminated the problems of the speakers and the distortion of the early order reflections that caused anomolies we then added another improvement to our medium: The Pressure Recording Process (PRP™ — trademarked by E. M. Long Associates) is a new type of microphone and miking technique developed by Ron Wickersham and Ed Long and manufactured under license by Ken Wahrenbrock, under the trade name PZM™ (trademarked by Syn-Aud-Con).

If you don't have an LEDE control room, or 813 monitors to A - B any mike in your arsenal, don't worry. The PZMs will give you a realism that will really make you a believer. Try it on anything. A - B it with any mike, in any situation — a piano is the most startling. Lay the PZM on a hard surface, on the floor, tape it to a baffle, tape it on a wall or under the lid of a piano — or tape **two,** one above the "f" holes and one down by the bass strings. This is the most incredible stereo piano you have probably ever heard. Horns, drums, vocals — all take on new realism. Some of the comments that I have received from some of the very fine trumpet players and musicians is that it is like playing against a solid concrete wall and hearing themselves come back, or a direct disk recording of themselves. Put your favorite vocal mike up for vocals and put the PZMs on a flat surface up to 3 or 4 feet away. (See Figure #3) Run one to one track, the other to

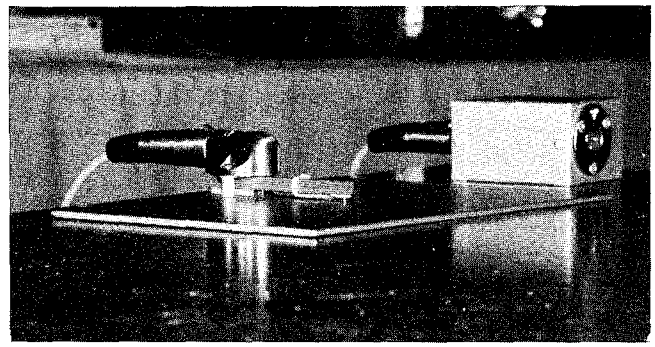


Figure 3. Pressure Zone Microphone (PZM™). A pair of Pressure Zone Microphones sells for $300 with power supply.

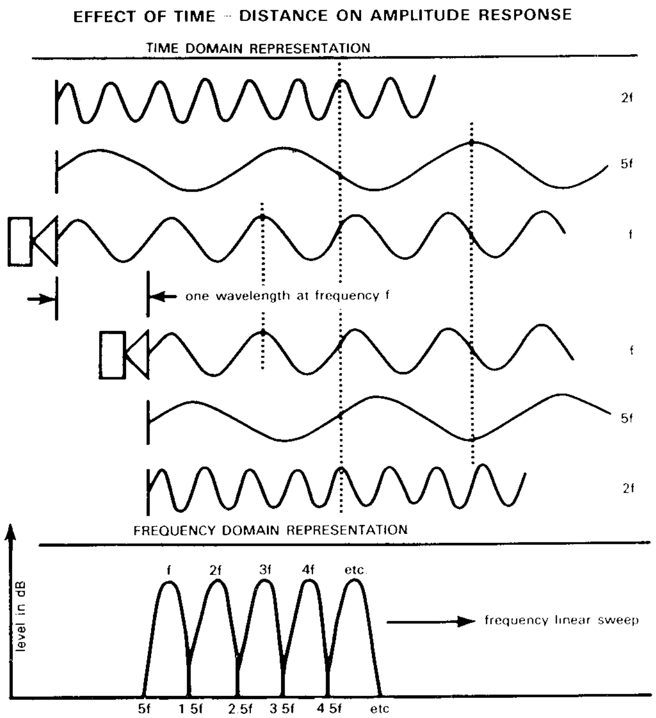
another. When you get through, A - B them. See which one the performer likes and see which one you like. If you don't believe the sound that you're hearing, go out in the studio and put your ear exactly where the mike is — exact realism.

Putting It All Together

When LEDE, TA, PZM and TDS are put together we have produced the most accurate mixing environment that we have had in the studio to date. It is so clear, and free of distortions caused by anomolies, mis-designed monitors, and general smearing of bad control rooms that your ears start to rebel. Where is that old sound of distortion, problems that everyone has accepted for many years? Your ears have learned a new realism that you can only experience from live performances — naturalness of drums, tympanies, trumpets, English horns, oboes. I have a good comparison for that — it's live orchestras, 40 weeks a year, seven days a week.

A large number of engineers have been through the new LEDE room at Las Vegas Recording. If you're near Las Vegas, I'll be happy to have you stop in.

What we have to look forward to now is an LEDE mastering room, digital recording and improved record processing. It is extremely difficult to put into words what these advancements can do for our industry, but I'll sum it up by answering the question that has been asked of me many, many times by the people who have come through our facilities: "What have you gained, if the material is to be played over AM radio, cheap hi-fi sets and television?" My answer to this has always been the same, "Every problem is additive, all problems add together — they never subtract from each other. They combine to make larger problems. Any problems that you can eliminate anywhere in the chain and make your product better will always be better, played anywhere, anytime, over any system." □ □ □



as yet, no reflections. Room modes are the resultant amplitude variations by frequency as measured at a given position of the complex additions of time varying reflected spectra with each other and the direct sound. In other words, "modes," or more properly, eigen-wavelengths (for it is their wavelengths that remain consistent as, for example, when the temperature shifts) are frequency response anomolies generated by reflected spectra. Figure #1 illustrates simple combinations that can occur. TDS allows the easy observation of this phenomenon because of the linear frequency response scale of TDS in distinction to the "log" scale universally used on standard frequency response charts. The anomolies generated by two spectra arriving at the same point in space are "out of phase," that is, displaced slightly in time, are linearly spaced nulls and peaks in frequency. See Figure #2.

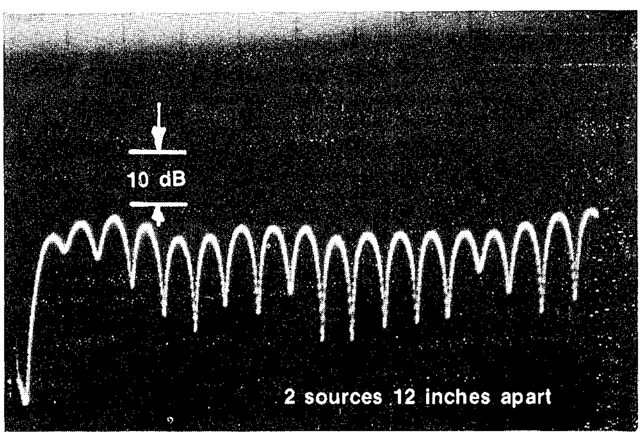


Figure 2: **The anomolies generated by two spectra arriving at the same point in space "out of phase", that is, displaced slightly in time, are linearly spaced nulls and peaks in frequency (0-20kHz)**

A Short Primer on Phase and Polarity

Many audio engineers accept the statement that "reversing the polarity of a speaker shifts its phase 180°."

If you examine the statement for a moment you will see that the signal emits from the **same point in space** with either polarity. The difference being that in one case it starts with a compression and in the other case it starts with a rarefaction. So let's clarify the terms "polarity" and "phase." (No Charlie — you don't "phase speakers" by reversing the connection to one of them.) Polarity **is not** frequency dependent; Phase **is** frequency dependent. A polarity reversal is instantaneous **in time** where a phase difference involves a time difference. In fact, there is a simple equation for translating phase difference into time difference

$$T_D = T_p\Theta / 360$$

Where:

T_D is the time delay, or difference, in seconds
T_p is the time period of frequency f ($T_p = 1/f$)
Θ is the phase delay in degrees

Then, of course,

$$\Theta = T_D/T_p\ (360)$$

For deliberate displacements of "acoustic center"
$T_D = ((1\ \text{sec}/1130\ \text{ft}) \cdot (1\ \text{ft}/12\ \text{in}))$ (displacement in inches)

Using these concepts we can calculate that two sources emitting the same spectra but with one of the units **one inch** behind the other.

$T_D = (1/1130)\,(1/12)\,(1'') = .0000737$ secs, or 73.7 μsecs.

The temporal integration window of the ear has an effective width of about 10 to 15 msecs. Therefore, it might be argued that such a slight time delay should not be audible. What is forgotten is what this "phase shift" (which is what time delay is) causes the amplitude response to do.

Figure #3 illustrates a frequency response from 0 to 20,000 Hz of the summation of the acoustic output of two loudspeakers (1" apart) at the measuring microphone. Please believe that the hole created is audible when program material falls in that region of the spectrum.

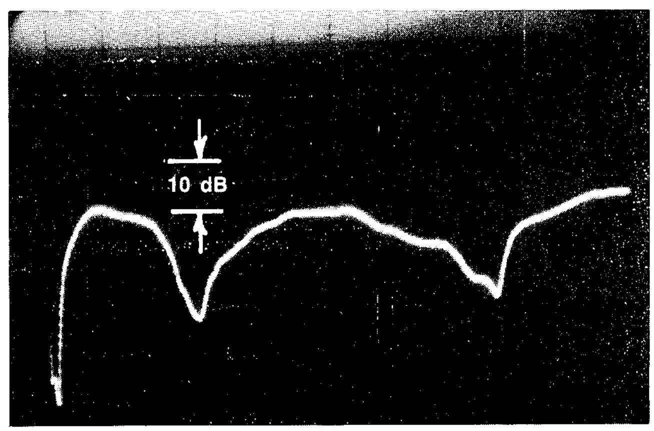


Figure 3: **Illustrates a frequency response from 0 to 20kHz of the summation of the acoustic output of two loudspeakers (1" apart) at the measuring microphone.**

Back To LEDE

Whenever the sound source and a pseudo source (such as a virtual image generated by a solid reflective surface) come together with a time difference of 73.7 μsecs or at a frequency of 6780 Hz, a phase difference of

$$\Theta = ((.0000737)/(1/6780)) \times (360) = 180°$$

Thus, you will have a large null in the response.

Now, let's place the two sources 10" apart. That is, one source reaches your ears 10" **ahead** of the other source. See Figure #4. We now have peaks at 1356, 2712, 4068, 5424,

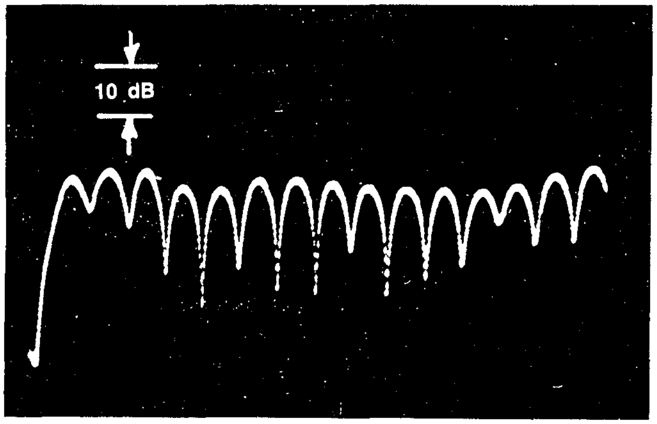


Figure 4: **Two loudspeakers 10" apart. One source reaches the ears 10" ahead of the other source.**

6780 and 8136 Hz, and nulls at 2034, 3390, 4746, 6102, 7458 and 8814 Hz.

What is of fundamental interest is the **fact** that the greater the time difference (or distance) between two combining spectra, the **narrower** the response anomolies generated. I'm sure we are all familiar with the concept that the wider the bandwidth of signals (at the same level) the greater the power. For example, one hundred watts as a sine wave will reach a far higher level on the analyzer than a 100 watt pink noise signal as the filter reads each spectrum. Thus the familiar caution, "the wider the bandwidth, the more likely to cause problems."

Only a short period of time is required before the investigation of response anomolies leads to an understanding that small time differences are not desirable in control room acoustics. How do we get rid of such early differences? **Put the sound source in as nearly anechoic space as you can achieve, but in a manner that insures that the reflected sound travels 15 to 20 feet further than the path taken by the direct sound from the source.**

Making a Physically "Small" Room Into An Acoustically "Large" Room

When the "source" end of the control room is made nearly anechoic it insures, so far as excitation by the sources located in the anechoic end are concerned, that the earliest reflection the ear will hear now has the time dimension of a much larger room. So, too, the diffusion at the "live" end is able to be manipulated so as to approach both the narrow band characteristics as well as the density characteristics of a much larger space than actually is on hand. In other words, all clues, acoustically speaking, have been removed or masked that would allow aural identification of the physical size of the environment. Beranek has written in **Music, Acoustics and Architecture** (page 26):

> Persons trained in listening — for example, blind people, who receive all their cues about the environment around them through the senses other than the eye — can "measure" the size of a room or judge the distance to a wall behind them **by the length of the time interval** between the **direct sound** and the **first reflected sound.**

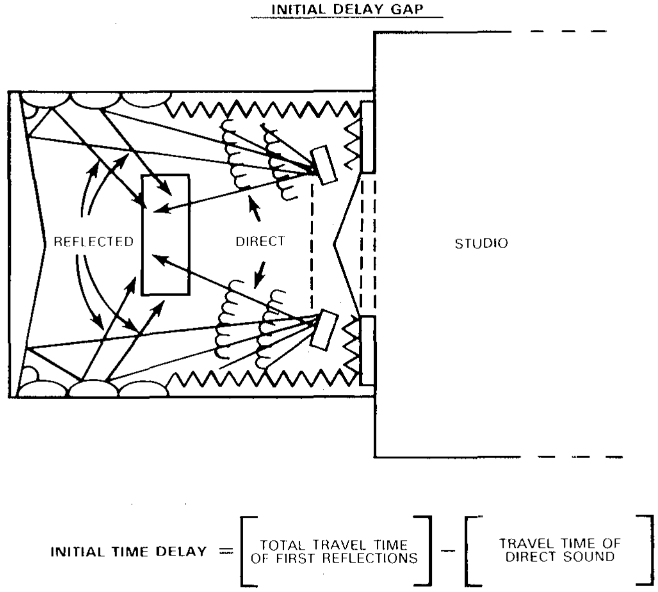
Beranek goes on to note that this capability is not restricted to the unsighted but that

> Experienced music listeners . . . sense the approximate size of a hall . . . by the length of the "initial-time-delay gap."

The LEDE technique, by virtue of the distance the direct sound must travel to encounter a **first reflection** has adjusted the initial time delay gap to the same figure that Beranek judged as desirable in the best concert halls in the world, namely 20 msec. It is no coincidence that the same 20 msec is the optimum delay for the maximum Haas effect in good diffuse semi-reverberant spaces. See Figure #5.

Figure 5

INITIAL DELAY GAP



There are two important factors both in the control room and in the concert hall. They are:

1 - The first substantial reverberant energy when in the Haas precedence zone does not distract the listener with "directional" information but merely raises the acoustic level.

2 - This early arriving reverberant energy must be well diffused if the aural mechanism is not to fasten on a specific clue. That is, the energy should arrive over a spread of time that does not allow identifiable broad band anomolies to be formed by only a few discrete reflections combining with extremely short differences in travel time. A mixture of a goodly number of reflections spaced over, say a time interval from 15 or 20 msec to 300 or 400 msec — the RT_{60} of the "live" end of the room — is the design goal in a really high quality control room.

All enclosed spaces **theoretically** have a reverberant sound level. Figure #6 illustrates the sound fields theoretically present in a control room. First, there is the direct sound level from the source. Second, there is the total reflected — reverberant sound level. This reverberant level may consist of merely a few discrete reflected spectra or it may be a well-established diffuse field. Third, there is the total sound level -- the direct and reflected sound levels combined. Finally, there is a sound field not generated by the sound source under consideration, called the ambient noise level. **All** of these sound fields require consideration when discussing any part of the total sound individually.

In control room work, because of a mistaken use of a technique useful in large rooms (arenas, etc.) but incorrect in small rooms, **many control rooms do not have a reverberant sound level capable of influencing the total sound level.** In other words, for all practical purposes there is no reverberant sound field present because the total sound field level is identical to the direct sound field level.

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DEFINITION OF ACOUSTIC TERMS

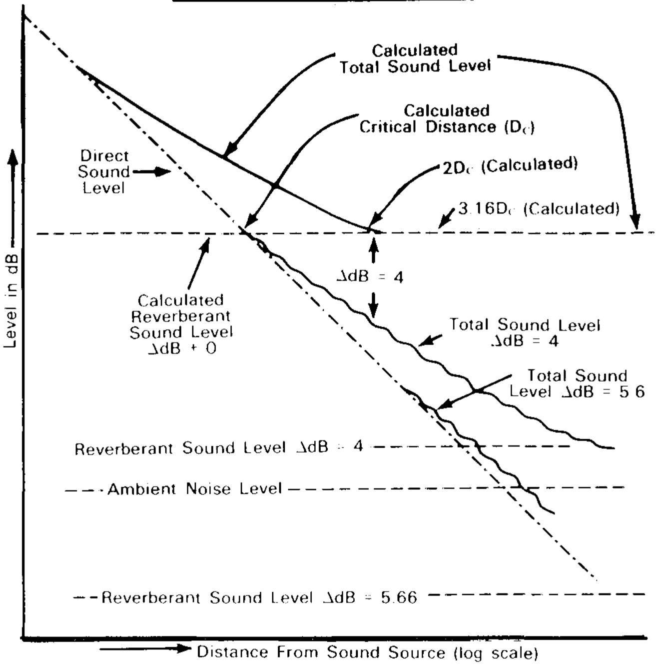


Figure 6

How A Control Room Can Be Anechoic To Its Monitor Loudspeakers While Remaining "Live" To The Mixer's Ears

First, consider if an anechoic chamber would make an acceptable control room. This question is easily answered by trying to mix in a chamber; a really horrible mix results. Or it can be seen theoretically by considering the massive anomolies that would be generated every time you moved your head from precisely between two monitors. Experience reveals that you wouldn't want to "mix" in a reverberation chamber either. Thus we have eliminated the two limits available. Obviously, the answer lies somewhere inbetween and is called the semi-reverberant sound field.

A semi-reverberant sound field is characterized by a reverberant level capable of influencing (favorably) the total sound level without having to be the predominant level. Figure #6 illustrates the effect.

One way to **thwart** the establishment of a semi-reverberant sound field that is quite often done without realizing the severe consequences is to make the wall behind the mixer "dead." See Figure #7. If, for example, there were a total absorption of 440 sabins ($S\bar{a}$) in the control room, so far as the loudspeaker is concerned, there are 440 x 64.9 = 28,556 Sa, while the mixer continues to hear a semi-reverberant space for any of the sounds he makes. At mid-frequencies a UREI 813 has a Q in excess of 10 (which, by the way, is nearly optimum — but that's a discussion longer than this article). A "live" talker has a $Q \simeq 2.5$ at mid-frequencies. Therefore, in this case the loudspeaker **will not generate a useable reverberant level.**

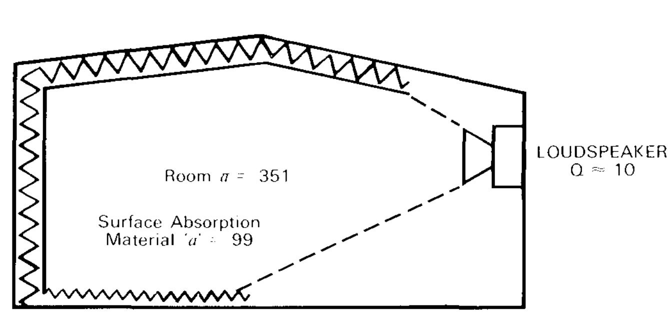
In large rooms (auditoria, arenas, etc.) we attempt to increase M_a (architectural acoustic modifier). In control rooms, we attempt to eliminate M_a and concentrate on maximizing diffusion in the "live" end of the room.

Some Comments On Reverberation

The above mentioned "diffusion" constitutes the heart of LEDE design and is a subject that, from the evidence in the literature of control room design, is not sufficiently understood. To properly cover the subject of diffusion requires more time and space than this article affords. For

Figure 7

EFFECT OF 'DEAD' REAR WALL



$$M_a = \left(\frac{1 - .351}{1 - .99}\right) = 64.9$$

If total $S\bar{a}$ = 440 then loudspeaker will generate a reverberant sound field level comparable to that which would appear in a space having $440 \times 64.9 = 28.556\ S\bar{a}$

those with pressing needs, the writings of A. M. Legendre and C. F. Gauss on quadratic-residue sequences of elementary number theory have been found germane by the most qualified of modern researchers into the problem.

Much confusion attends the term "reverberant sound field" in the control room. Theoretically speaking, any enclosed space has a reverberant sound field. The only difference between one space and another space is the **sound level** of the reverberant sound field.

To again look at "limits," an anechoic chamber has a "reverberant sound field" that is, by specification, down 20 dB below the direct sound field level. In any practical case, using the best of the currently available acoustic measurement equipment, an accuracy of 0.5 dB is exceptional; therefore, any reverberant sound field level found to affect the **total sound field level** by less than 0.5 dB may, with justification, be treated as not present. This is particularly true whenever the measurement of the total sound field level is being taken at a distance from the sound source that exceeds twice the calculated critical distance, D_C. See Figure #8.

We have found that the desired ΔdB falls between 3 and 4 dB. An extremely simple but highly effective estimator of ΔdB is the Peutz equation

$$\Delta dB = .22\ ((\sqrt{V})/(h \cdot RT_{60}))$$

Where:

V is the internal volume of the space in ft^3

h is the height of the ceiling

RT_{60} is the reverberation time for 60 dB of decay

We use the equation in the form of

$$RT_{60} = .22\ ((\sqrt{V})/(h \cdot \Delta dB))$$

Advantages of the LEDE Concept

What LEDE accomplishes acoustically can be enumerated as:

1 - Frequency response anomolies generated by the interaction of the direct sound and first reflections are high density narrow band (less than 1/6-octave at 500 Hz and 1/12-octave at 1,000 Hz, etc.).

2 - The initial time delay gap (as defined by Beranek) may be adjusted to provide the psychoacoustic effect of a large space.

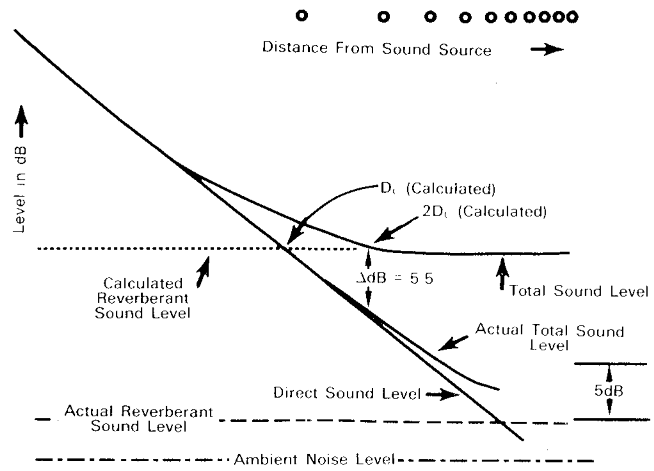


Figure 8

Figure #8: Sound Levels

The sound levels normally measured with a sound level meter are:

1 - Direct sound level

2 - Reverberant sound level

3 - Ambient noise level

4 - Total sound level.

Expressed mathematically the non-coherent addition of the first three constitutes the fourth.

$$T_{SL} = 10 \log (\exp(D_{SL}/10) + \exp(R_{SL}/10) + \exp(ANL/10))^{*}$$

The addition of individual frequencies can be accomplised by

$$T_{SL} = 10 \log \sqrt{(\exp(SPL_1/10)^2 + (\exp(SPL^2/10))^2 + 2(SPL_1)(SPL_2)(\cos(a_1 - a_2)}^{*}$$

* exp = base 10

Where SPL_1, SPL_2 is the relative sound pressure associated with a sound level

$$10(D_{SL}/20) = SPL_{rel}$$

a_1; a_2 are the phase angles in degrees associated with SPL_1 and SPL_2.

It is important when using sound level meters, spectrum analyzers, etc., that the operator knows how to detect which of these levels or combination of levels he is measuring and knows how to calculate the others from the ones measured. Quite often the total sound, including ambient noise, is mistaken for the reverberant sound level. On occasion critical distances are discovered in essentially non-reverberant spaces and incoherent signals are summed as coherent signals.

Figure #8 helps define some of these terms and illustrates their relationship. The case chosen is a control room which for all practical purposes may be considered without a reverberant sound field. The actual reverberant level is so low as to make the total sound level at 2D$_C$ (calculated) less than .5 dB different than the direct sound level at that same distance. Since the room is only 40 feet in its longest dimension, the reverberant sound level, though theoretically present, does not affect any measurement undertaken.

3 - All first order reflections fall within the Henry-Haas effect precedence zone.

4 - Substantially more acoustic energy can be developed without incurring the penalty of "small space" coloration.

5 - Symmetry need not be as rigorously enforced provided no repetitive reflection paths are established (control of flutter, echoes, etc.).

6 - A sufficiently reverberant sound field may be developed to generate a ΔdB of 3 to 4 dB while maintaining traditionally accepted reverberation times.

7 - Spatial geometry is not degraded by the control room environment and therefore remains dependent upon:

A - Loudspeaker spacing and orientation (10 to 20 feet are suggested separation distances).

B - Time alignment of loudspeakers.

C - Pressure Zone Microphony, PZM™. Suggested microphone spacing just slightly wider than playback loudspeaker spacing used.

D - One fundamental advantage of LEDE that can be overlooked is the fact that equipment racks, tape machines, etc., when placed in a soffit flush with the upper rear wall now provide **useful** diffusion of the higher frequency energy.

Conclusions

The LEDE concept of control room design is part of a chain of events that leads to "coherent sound reproduction."

1 - TDS analysis of direct and reflected spectra.

2 - Enlightened use of actual Time Aligned™ (E. M. Long Associates) transducers. (Be very wary here as a majority of the devices claiming TA are not so aligned.)

3 - Enlightened use of the PZM™ system.

4 - Creation of a desirable "acoustic size" control room free of uncontrolled response anomolies.

5 - Creation of a diffuse "live" end adjusted to an effective ΔdB.

6 - Use of combinational geometry in the creation of eigen wavelengths of the desired spacing and density.

7 - Use of the psychoacoustic effects of precedence (Henry, Fay-Hall, Haas, et al) and intitial time delay gap (Beranek).

Chips Davis is the first recording engineer to use PZM™, TA, and LEDE, in a control room analyzed by TDS. □ □ □