"SYN-AUD-CON"
SYNERGETIC AUDIO CONCEPTS

P.O. BOX 1134, TUSTIN, CALIFORNIA 92680

newsletter

VOLUME 5, NUMBER 1
OCTOBER, 1977

EDITORS: Don & Carolyn Davis

SYNERGETIC
Working together; co-operating, co-operative

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

EXCHANGE OF IDEAS

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

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

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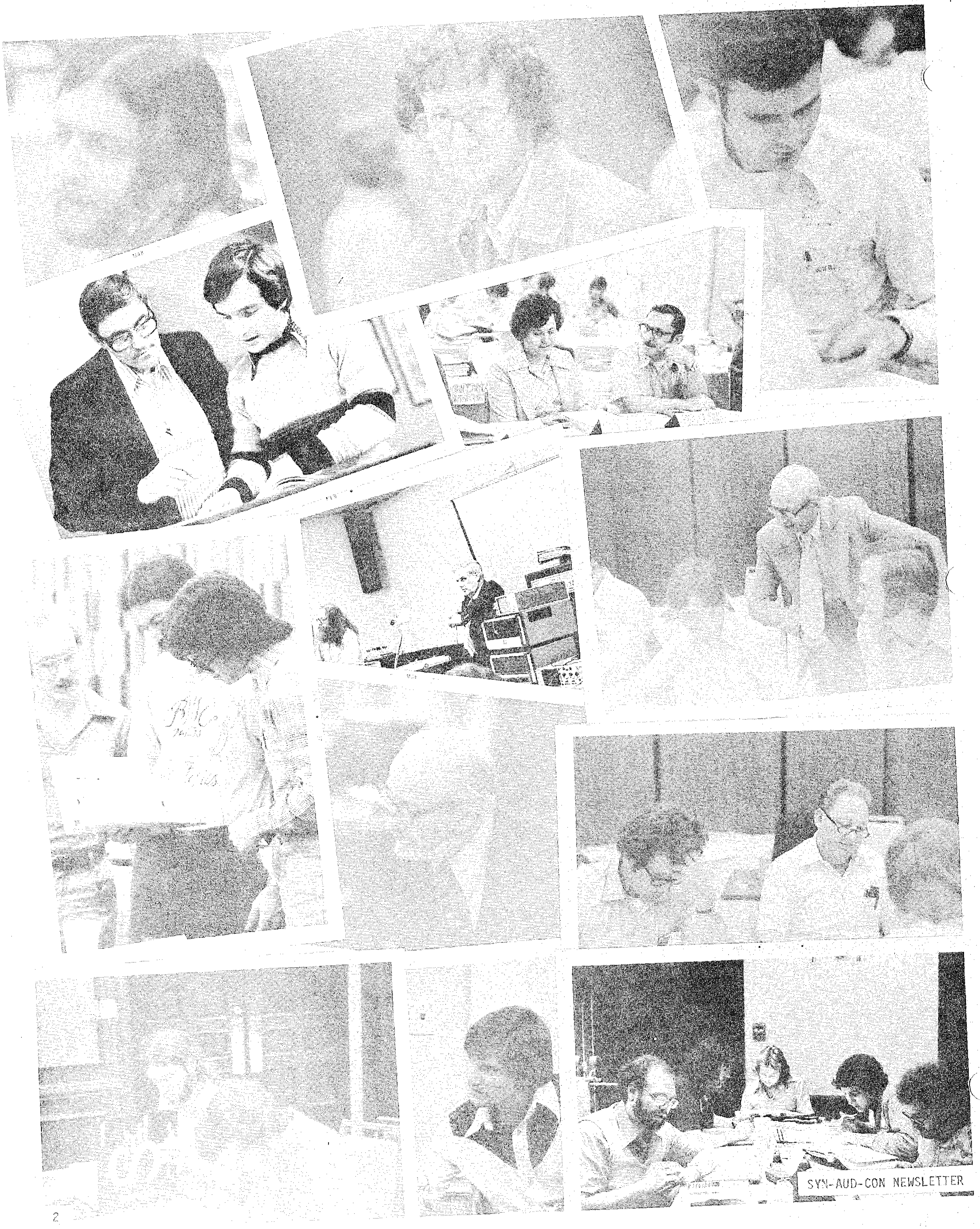


DRUMMER ISOLATION

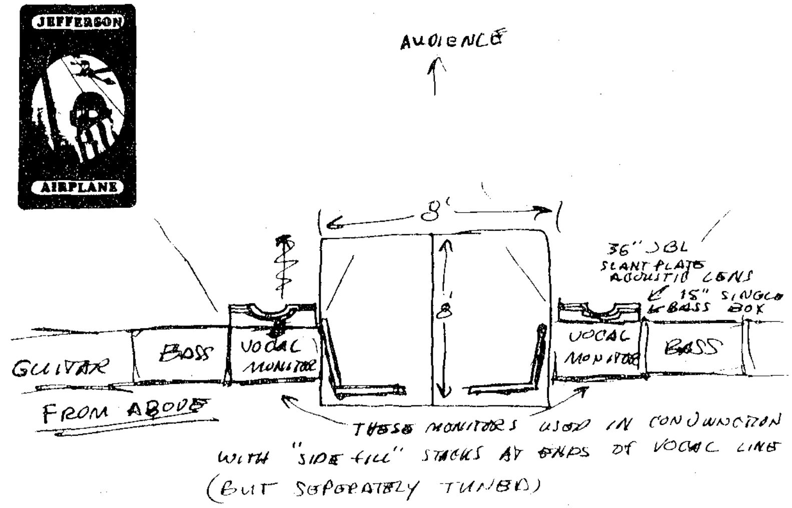
NYYA LARK (2-time DC graduate) wrote last year asking a couple of questions, one of which we didn't feel equipped to answer. JIM COE in our class in San Francisco in 1975 said that he would write Nyya.

Nyya's question: I've seen a number of times on television different drummers with headsets on, used in place of monitors. I've also seen a conductor of a television orchestra using the headsets. They also had regular floor monitors and they were being mixed on the studio floor. My question is, can such a method be done for the drummer during live shows on the road? and How? (ED's Note: Nyya was a mixer and sound technician for Gil Scott-Heron at the time she wrote)

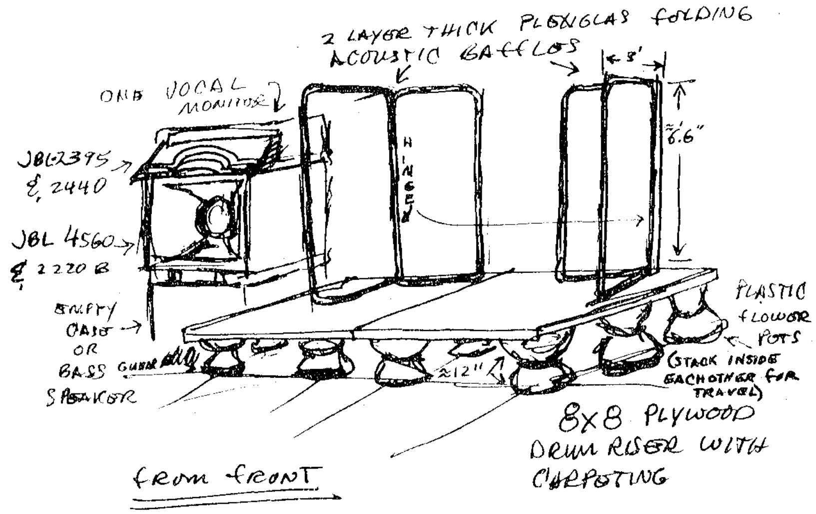
Jim Coe's answer:

Carolyn Davis asked me to give you some advice in answer to your recent letter, since I was in the S.F. Seminar. For the last five years, I've been head "soundman" for the Airplane, Hot Tuna and now the Starship. Now to try to answer your question.

Many Drummers have trouble hearing both their drums and the vocal monitors in loud bands. The reasons are usually distance from the monitors and masking of the drums by louder guitar and/or bass amps nearby. Headphones are one way to go, but they seldom sound natural and may isolate too much or little. Their signal will be out of phase with any sounds coming from more than a few feet away unless (expensive) time delay is used. For the least isolation I recommend the Sennheiser Model 414 "open air" phones. For the most isolation I use (for soloing Mics and busses at the mixer) 414's removed from their headbands and epoxied into David Clark noise suppressors such as you see on jet airport ground crews. This gives about 20-30 dB isolation.



For our drummers we solve the problem acoustically with a double layer Acrylic plastic baffle/reflector set standing behind the drummer and a risor of Marine plywood with Varathane coating covered with carpet. The baffle/reflectors isolate the drummer somewhat from the instrument speakers near him and reflect his sounds toward the front rows, where P.A. coverage is weakest near the center.



The risor raises the drummer above the "ground plane" of the stage. We have acoustic consultant Peter Quaintance of S.F. to thank for this idea and equipment. It works so well that we stopped having to feed the drummer a "drum mix" in a special monitor (an attempt doomed to failure by the complex phase interactions of the direct sound from the drums and the monitor signals, to say nothing of the strange leakage problem in the house system).

Our present Drummer needs no special monitor at all, since he is only a few feet from the vocal monitor acoustic lens end. (check the drawing). We use vocal monitors on the sides and in the amp line. This allows the guitarists to hear when they are very near their amps as well as near the vocal mic line.

THE COURT SWITCHES FRANCHISING SIGNALS

Quoting from BUSINESS WEEK: The high court threw out an antitrust rule it had handed down exactly a decade ago. At that time the justices told Arnold Schwinn & Co. that when the company sold bicycles to an independent middleman, it could not limit the middleman's sales to a particular territory. Such restraints on competition, the 1967 ruling said, are per se illegal under the Sherman Antitrust Act, meaning that there is never any justification for them. That is the approach that the court now says was a mistake. (ED'S NOTE: It took them 10 years to find out what we all knew.)

The Schwinn per se rule "was a sword that hung over everyone's head," explains Anthony J. Obadal, lawyer for Associated Equipment Distributors. Dealers in a particular territory could not be assured that those from elsewhere would not steal away customers, and some lawyers even had clients write into franchise contracts a clause saying that there was no assurance of territorial exclusivity.

One company that will not be using the new freedoms given manufacturers is Schwinn: After the 1967 ruling, it got rid of all its independent distributors and established a wholly owned subsidiary to handle sales to retailers. (Excerpts from article)

IMPEDANCE TESTER

by
Glen Ballou

The Impedance Tester, as shown on page 114 of *SOUND SYSTEM ENGINEERING* can be expanded for easier operation. By installing calibrating resistors in the tester, the Real Time Analzyer output can be easily adjusted for each standard impedance.

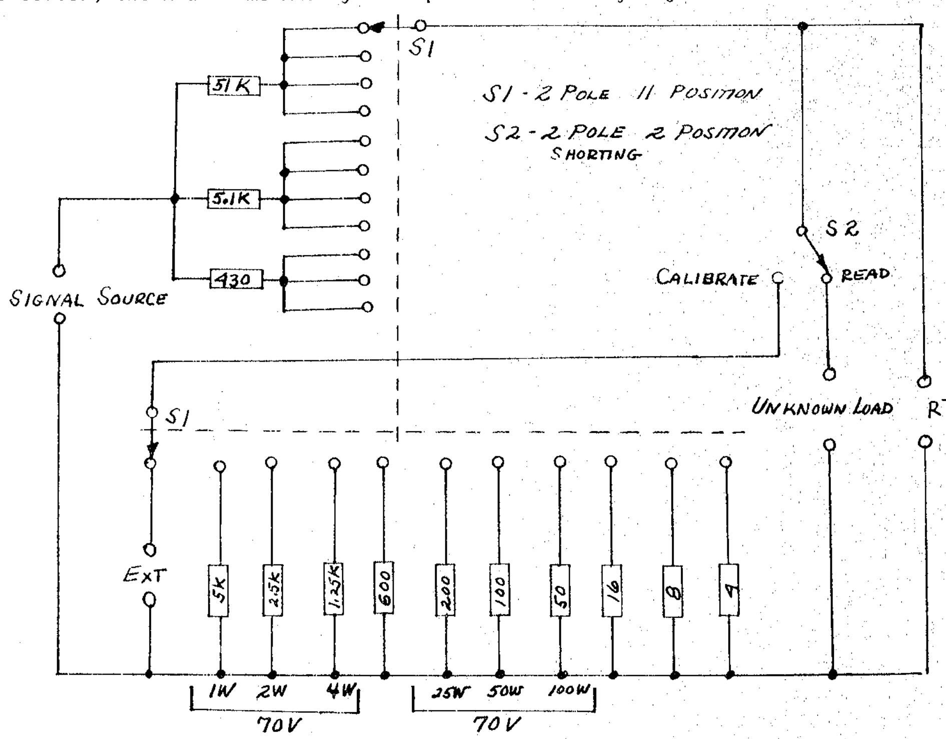
The theory of operation is simple. In "Calibrate" a voltage divider is made of the 51KΩ, 5.1KΩ or 430Ω resistor and the equivalent load resistor (5KΩ - 4Ω). By observing the junction of the voltage divider and common on the RTA, we see a staight line which can be adjusted for the reference impedance.

Switching the Calibrate-Read switch replaces the calibrated resistor with the load. If the load is purely resistive, and the same resistance as the standard, the RTA scan will be the same. If the impedance is twice or one half the standard, the RTA will be ±6 dB from the standard.

$$Z_{unknown} = (10^{\Delta dB/20})\ Z\ standard$$

If the load is not a pure resistance, the scan will vary up and down around the standard, depending on the impedance at each frequency.

Due to the high impedance of the box, 1/2 watt resistors can be used.



If you make a S-1 a 3-pole 11 position switch, then the calibrating resistor can be mounted across two wafers of the switch, simplifying construction.

MORE ON TDS

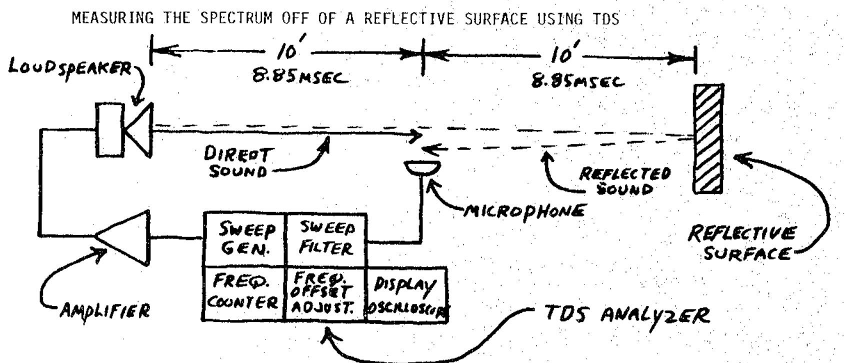
Recent classes have told us that Time Delay Spectrometry would be easier to understand if they had a typical equipment hookup diagram to look at while studying the diagram on Page 3 of the July Newsletter (Vol 4 # 4).

Figure # 1 illustrates the equipment used and its general hookup for measuring the spectrum of a reflection (ignoring the direct sound spectrum while doing so). The sweeping generator is an HP 8556A (cost $2450). The sweeping receiver is an HP 8552B (cost $3775.). The display oscilloscope is an HP 141T (cost $2325) The counter is a 5304A Timer (cost $385) plus an HP 5300A measuring system (cost $500). The frequency offset adjustment is a custom modification done under license to Cal Tech.

The sweep rates (SR) are linear, i.e. 10,000 Hz/sec or 10 Hz/msec. The frequency offset calculated remains constant between the generator and receiver throughout the sweep with the generator leaving 0 Hz, in our example 265.49 Hz ahead of the filter and finishing the sweep at 10,000 Hz, 265.49 Hz ahead of the filter.

See Page 8 of this Newsletter for photographs of actual TDS results.

MEASURING THE SPECTRUM OFF OF A REFLECTIVE SURFACE USING TDS



FREQ. OFFSET ADJUST. SETS SWEEP FILTER TO BEGIN SWEEP AFTER SWEEP GEN. REACHES

$$S.R.(UFL - LFL)\left(\frac{\text{TOTAL DIST. REFL. SOUND TRAVELS TO MIC}}{\text{VELOCITY OF SOUND}}\right)$$

IN CASE SHOWN ABOVE THE SWEEP FILTER BEGINS SWEEP FROM 0 HZ (S.R. = 1 SEC, UFL = 10,000 HZ, LFL = 0 HZ) WHEN GEN PASSES:

$$1\,sec(10{,}000 - 0)\left(\frac{30}{1130}\right) = 265.49\ Hz$$

STEVE SIMPSON, JR (2-time graduate) of Southwest Sound & Electronics, Inc., San Antonio, TX, is not one to mince words. More importantly he has the experience and knowledge to back up what he has to say. His Letter to the Editor of YOUR CHURCH is powerful and it is reproduced below as an excellent example of what is amiss in church building and what a church sound committee can do to avoid the major pitfalls. Steve makes many points that every sound contractor in the business of selling church sound systems can use.

LETTER TO "YOUR CHURCH"

FROM
STEVE SIMPSON, JR

Dr. Lyle Schaller
Mr. Richard L. Critz
c/o YOUR CHURCH
The Religious Publishing Company
198 Allendale Rd.
King of Prussia, PA 19406

Gentlemen:

Your article in the September/October 1977 magazine YOUR CHURCH was read with interest from the standpoint of a sound contractor that spends a major portion of his time helping congregations hear what the preacher is saying.

I think you gentlemen have missed one of the more important problems with the modern church today.

The older churches - even the small ones - usually were built for the preacher with a great, booming voice. The new churches that are being built largely in modern designs are, a large percent of the time, designed without an architect employing a sound consultant to see that the church will "talk" satisfactorily. In fact, most churches are built without specifying or considering a sound system which is left until the last when the preacher has a beautiful new installation but no one can hear. He now discovers that it is going to cost from 10 - 50% more to put a sound system in the church than it would have cost had it been planned in the original design.

It is not unusual for me to visit an older church in which either the Chairman of the Building Committee, the Minister, or the Director of Music brings up the status of the church by explaining that they have just spent $10,000 on new carpet, $5,000 on new lighting and $7,000 on pew cushions. Then they ask me to provide a good sound system. When I advise them that the price will be from $4,000 - $6,000, they are shocked for they expected it would cost in the order of $1,500 - $2,000. My comment: "YES SIR, YOU HAVE JUST SPENT APPROXIMATELY $21,000 FOR THE COMFORTS OF YOUR CONGREGATION SO NOW ALL THEY CAN DO IS TAKE A COMFORTABLE NAP DURING THE SERVICES BECAUSE THEY CAN'T HEAR." (ED:Caps mine)

I won't go into the problems of "You can't mount speakers on my beautiful ceiling - You can't put a central cluster of horns in the peak of the ceiling because it won't look nice" or "They block the chandeliers or accoutrements." Most of these problems can be overcome by compromise or an agreement on use of materials by the sound contractor and the architect.

We have learned a lot about acoustics, sound absorption, and sound reinforcing in the last 10 years to the extent that a good, reputable sound contractor can guarantee the required sound pressure level to every seat in the house, based on your requirements, or remove the system at his expense. You won't find any reputable sound contractor that isn't sure enough of himself to risk the removal of an expensive sound system at a very serious loss.

You mentioned casually that there were some problems with hard of hearing people wanting to participate in the regular services instead of being placed in the equivalent of the cry-room. There are a certain percentage of hard of hearing in every congregation, some of which are ashamed to admit they are hard of hearing and don't want to use a hearing aid. There are new methods, types of equipment, and techniques and taking care of the hard of hearing should be anticipated.

Since I am in the business of providing high quality, equalized sound systems with a guarantee that my installations will work satisfactorily, I, of course, have an ax to grind in my above comments. I would like to make the following suggestions that should be added to your next article concerning "The Future of the Church in America."

1. In building a new church, insist that your architect include consultations with either a sound consultant or a sound contractor who is a professional and can provide both design and installation with a guarantee.

2. Instruct the architect that there will be a member of the Building Committee who will consult with the architect and the sound consultant to be sure that the type of the system and the aesthetic effects, both archicectural and aural, are agreeable, and that the additional services, such as provisions for hard of hearing, cry room, overflow room, narthex, and outside sound are taken care of. Be sure that it is distinctly understood whether you wish a "classical speech only" sound reinforcing system which is fairly economical, or do you wish the more "modern church" technique, where entertainment, music and theatrical type of performances are to be covered by the sound system.

3. In old installations, be sure that you are negotiating with a sound contractor and/or consultant who is capable of calculating or measuring the important parameters which control a sound system such as: room absorption, reverberation time, whether there is a need for time delay for long rooms, articulation loss of consonants, and background noise from electrical equipment, mechanical equipment, or traffic noise. A person who is capable of measuring or calculating the above would have the several thousands of dollars worth of measuring equipment needed to provide a satisfactory installation.

4. If you do not want to go the route of employing a sound consultant who will design your system and charge you for the design, which is then put out for bid, you can negotiate with a sound contractor who is also a professional and who will provide you with a high quality job with a guarantee.

...continued next page

Steve Simpson, Jr - Letter to the Editor, continued

5. Be sure that you do not become a peddler of bids by asking for written design information from several sound contractors and then passing along the engineering information to their competitors because if the sound contractor is a first class, reputable person, he has spent many hours and possibly hundreds of dollars designing a top notch system for you.

It doesn't do much good to do all of the things that were proposed in your article when your "customers can't hear about what you are selling."

Sincerely,

S. H. Simpson, Jr.

SOUND ABSORPTION COEFFICIENTS

Sound absorption coefficients of acoustically efficient absorbing materials.

Description	frequency range of normal speech Hertz 125	250	500	1000	2000	4000
Acoustical plaster ("Zonolite") ½", trowel application	31	32	52	81	88	.84
same 1" thick	25	.45	.78	.92	89	.87
Acoustile surface glazed and perforated structural clay tile perforated surface backed with 1" glass fiber blanket of 1 lb ft density	.26	.57	.63	.96	.44	.56
Fiberboards ½" normal soft, mounted against solid backing, unpainted	.05	.10	.15	.25	.30	.30
same painted	.05	.10	.10	.10	.10	.15
½" normal soft, mounted over 1" air space, unpainted	.30		.15		.10	
same, painted	.30		.15		.10	
Fiberglass insulation blankets AF100, 1", mounting #4	.07	.23	42	.77	.73	.70
same, 2", mounting #4	19	.51	79	.92	.82	.78
AF530, 1", mounting #4	09	.25	.60	.81	.75	.74
same, 2", mounting #4	20	.56	89	.93	.84	.80
same 4", mounting #4	.39	.91	99	.98	.93	.88
Flexboard, 3/16" unperforated cement-asbestos board, mounted over 2" air space	.18	.11	.09	.07	03	.03
Geocoustic, 13½" x 13½", 2" thick cellular glass tile, installed 32" o c , per unit	.13	74	2.35	2.53	2 03	1.73
Hardboard panel ⅛", 1 lb/ft with bituminous roofing felt stuck to back, mounted over 2" air space	90	.45	.25	.15	.10	.10
Masonite ¼", mounted over 1" air space	.12	.28	.19	.18	.19	.15
Mineral or glass wool blanket, 1", 5-15 lb/ft density, mounted against solid backing covered with open-weave fabric	.15	.35	.70	.85	.90	.90
same, covered with 5% perforated hardboard	.10	.35	.85	.85	35	.15
same, covered with 10% perforated or 20% slotted hardboard	.15	.30	.75	.85	.75	.40
Mineral or glass wool blanket, 2", 5-12 lb/ft density, mounted over 1" air pace, covered with open-weave fabric	.35	.70	.90	.90	.95	.90
same, covered with 10% perforated or 20% slotted hardboard	40	.80	.90	.85	.75	
Plywood panels 2", glued to 2½" thick plaster wall on metal lath	.05		.05		.02	
¼", mounted over 3" air space, with 1" glassfibre batts right behind the panel	.60	.30	.10	.09	.09	.09

Sound absorption coefficients of acoustically efficient absorbing materials.

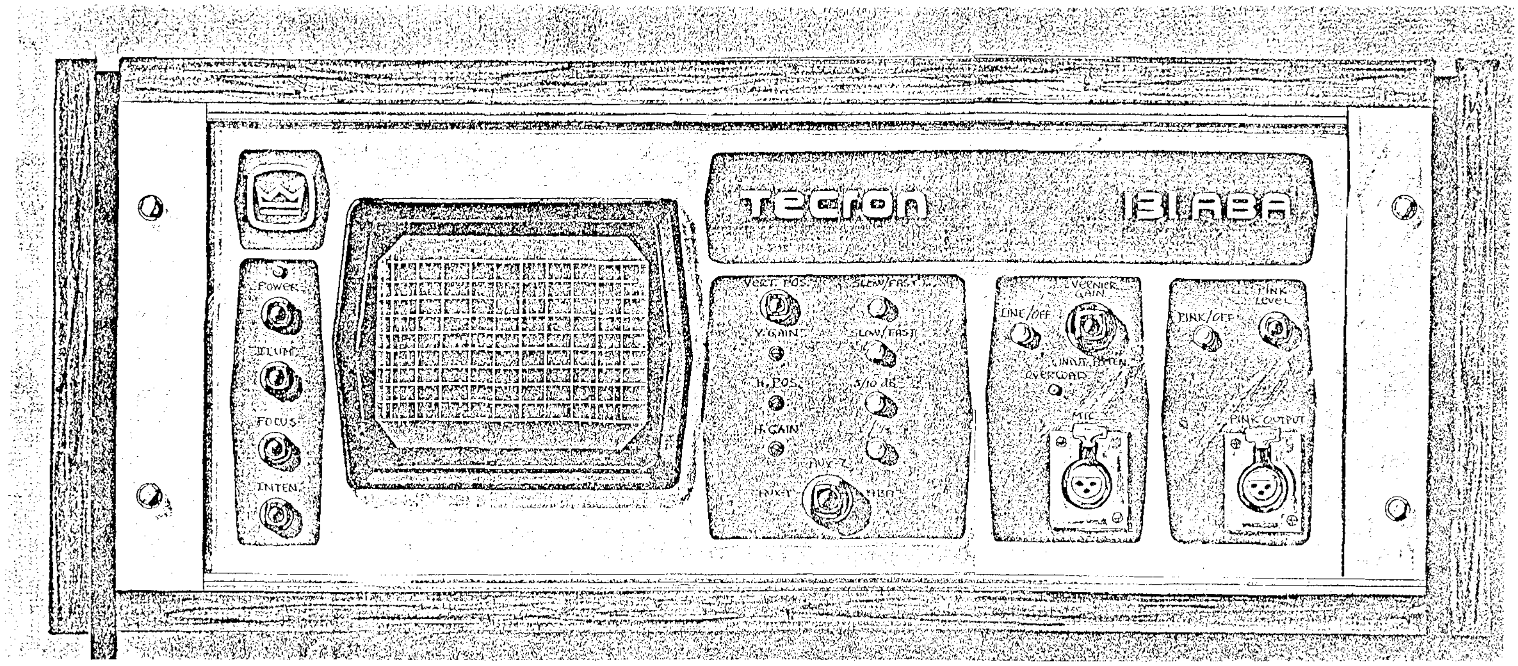
Description	frequency range of normal speech Hertz 125	250	500	1000	2000	4000
Rockwool blanket, 2" thick batt ("Semi-Thik"), mounted against solid backing	.34	.52	94	.83	.81	.69
mounted over 1" air space	.36	.62	.99	.92	.92	.86
mounted over 2" air space	.31	.70	.99	.98	.92	.84
Rockwool blanket, 2" thick batt ("Semi-Thik"), covered with 3/16" thick perforated cement-asbestos board (Transite), 11% open area, mounted against solid backing	.23	.53	99	.91	.62	.84
mounted over 1" air space	.39	.77	99	.83	.58	.50
mounted over 2" air space	.39	67	.99	.92	.58	.48
Rockwool blanket 4" thick batt ("Full-Thik"), mounted against solid backing	.28	.59	.88	.88	88	.72
mounted over 1" air space	.41	.81	.99	.99	92	.83
mounted over 2" air space	.52	.89	.99	.98	.94	.86
Rockwool blanket, 4" thick batt ("Full-Thik"), covered with 3/16" perforated cement-asbestos board (Transite), 11% open area, mounted against solid backing	.50	88	.99	.75	.56	.45
mounted over 1" air space	.44	.88	.99	.88	.70	.30
mounted over 2" air space	.62	.89	.99	.92	.70	.58
Roofing felt, bituminous, two layers, 0 8 lb/ft², mounted over 10" air space	.50	.30	.20	.10	.10	.10
Spincoustic blanket 1", mounted against solid backing	.13	.38	.79	.92	.83	.76
2", mounted against solid backing	.45	.77	.99	.99	.91	.78
Spincoustic blanket, 2" thick, covered with 3/16" thick perforated cement-asbestos board (Transite), 11% open area	.25	.80	.99	.93	.72	.58
Sprayed "Limpet" asbestos ¾", 1 coat, unpainted on solid backing	.08	.19	.70	.89	.95	.85
same, 1" thick	.30	.42	.74	.96	.95	.96
¾", 1 coat, unpainted on metal lath	.41	88	.90	.88	.91	.81
Transite, 3/16" perforated cement-asbestos board, 11% open area mounted against solid backing	.01	.02	.02	.05	.03	08
mounted over 1" air space	.02	05	.06	.16	.19	.12
mounted over 2" air space	.02	.03	.12	.27	.06	.09
mounted over 4" air space	.02	.05	.17	.17	.11	.17
paper-backed board, mounted over 4" air space	.34	57	.77	.79	.43	.45
Wood paneling, ⅜" to ½" thick, mounted over 2" to 4" air space	30	.25	.20	.17	.15	.10

LOWELL T. COURT (Alberta Extension class in Edmonton) of Allsopp, Morgan Engineering Ltd., Consulting Engineers, Edmonton, Alberta, sent the tables above to us. They appeared in Canadian Interiors in September 1975. It is always with great interest that we add to our tables of absorption coefficients.

NEW REAL TIME ANALYZER FROM CROWN

Advanced data from CROWN INTERNATIONAL on their real time analyzer came just in time to be included in the Newsletter. Crown hopes to ship units by January-February 1978. The unit we saw at the Spring AES convention worked beautifully.

The 131 ABA is an excellent buy in its price category of approximately $2500 (retail!) and far exceeds the capabilities of the older HP-Altec 8050A.



Special features that are most desirable

1. Built in pink noise generator
2. 55 dB dynamic range
3. 16 to 20,000 Hz frequency response
4. Selectable 1 and 1/3 octave filters
5. Easily photographed 5" display
6. Dual averaging speeds
7. Low, low price - $2500 (subject to change)

The Crown 131 ABA illustrates one of the exciting aspects of working with Syn-Aud-Con sponsors. It was just a year ago (1st of November) that we held a special class at Crown for 40 of their factory people and representatives. Their EQ-2 equalizers were new then and we spent some time tuning with the real time analyzers we had with us and we discussed some of the difficulties of tuning with a record. It wasn't more than a few months later that Crown told us that they would have a RTA in the product line.

On the subject of the EQ-2, many keen ears can hear an active filter in the circuit without a single filter being turned in. For that reason some recording studios will not use speaker-room equalization. The EQ-2 system is not detectable in the circuit.

WHERE CONSTANTS COME FROM

DAVID LYNCH 2-time graduate from San Diego asked an interesting question in the May Los Angeles class.

In converting open circuit voltage readings in dBV/1 dyne/1 volt into power available readings in dBm, the equation reads

$$\text{dBm rating} = \text{dBV rating in dB} - 10 \log Z + 44$$

The question was "where does the +44 come from?"

First of all, the dBV rating is the number of dB below 1 volt. 1 volt across 1Ω = 1 watt

The dBm rating is referenced to 0.001 watt.

$$10 \log \frac{1 \text{ watt}}{0.001 \text{ watt}} = 30 \text{ dB}$$

Therefore, if we changed *only* the reference point we would have a reading 30 dB closer to reference or +30 dB.

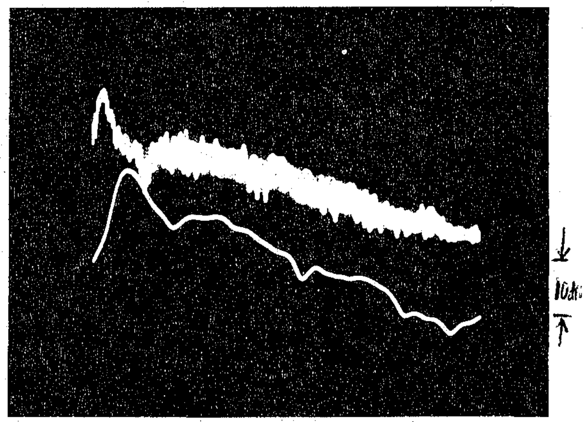
Secondly, in obtaining the dBV rating, we input the microphone with 74 dB-SPL. In obtaining the dBm rating, we input the microphone with 94 dB-SPL. Therefore, if we *only* changed the acoustic input level we would be 94-74 = 20 dB higher rating.

Finally, in making the dBV rating, it's the *open circuit* voltage whereas in the dBm method, it is adjusted 6 dB to account for theoretical termination. Therefore we subtract 6 dB: 30 + 20 - 6 = 44.

LOOKING AT THE DIRECT-TO-REVERBERANT RATIO

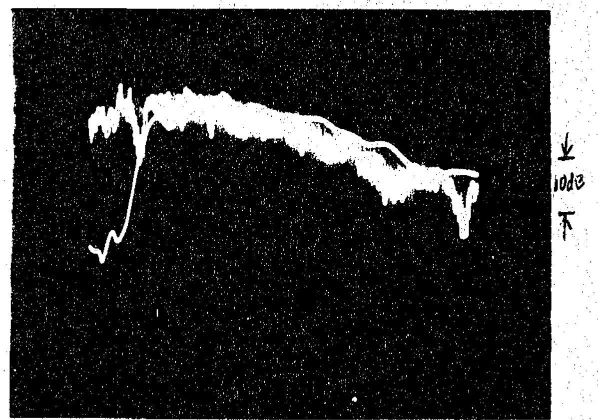
Until the advent of time delay spectrometry (TDS) views such as shown here were nearly impossible to obtain with any kind of economy of either time or money.

#1



1000 TO 10,000 HZ

#2



1000 TO 10,000 HZ

Curve # 1 illustrates our regular loudspeaker used in class (Rauland MLS-3) at a distance well beyond D_C. The upper spectrum curve is the *total* sound field at the microphone and the lower spectrum is the *direct* sound at the microphone.

In this case the total sound spectrum is almost all reverberant sound. It is interesting to note the similarities of the two spectrums. The direct sound is more than -10 dB *below* the total sound field and therefore is making only a small contribution to it.

Photograph # 2 is of a measurement of the large EV HR 4020 horn made at the same distance as Photograph # 1. It's still under D_C at the same distance that the other unit was -10 dB beyond D_C. Since TDS shows direct sound and then the total sound spectrums, it's not possible to view only the reverberant sound field when at or under D_C.

TDS allows remarkably detailed analysis of direct sound spectrum behavior at distances much greater than D_C and gives us our first overall picture of what our ears are hearing in real life spaces.

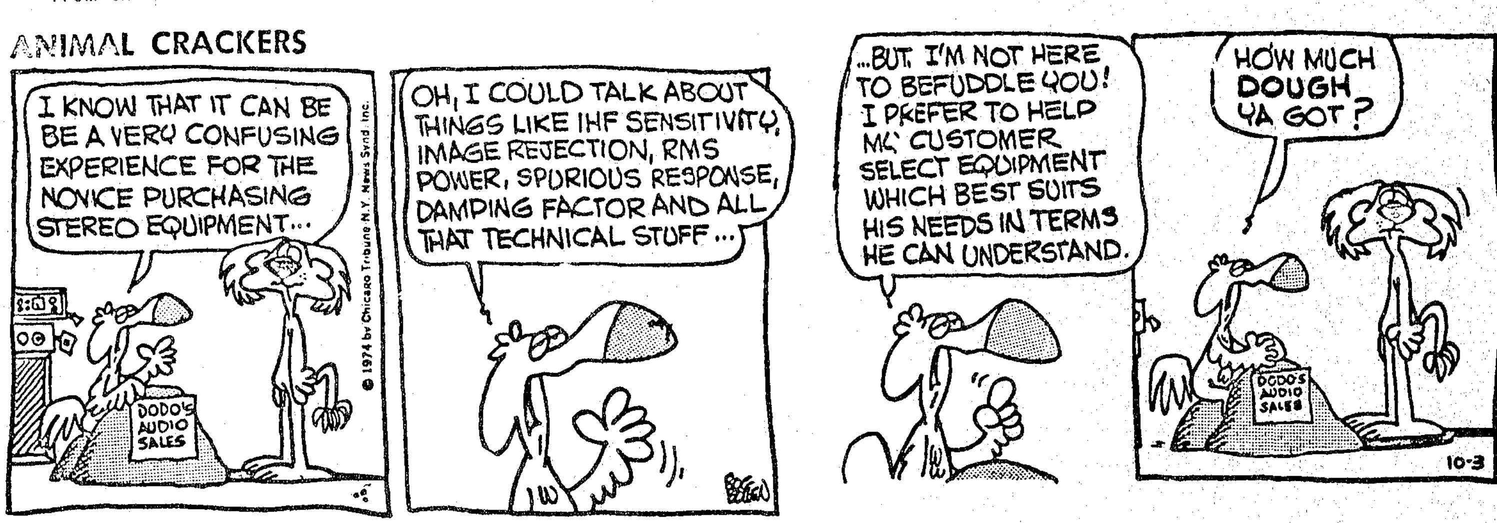
Of particular interest to us is the fact that the direct-to-reverberant ratio stays so uniform over such a wide portion of the spectrum.

SIMPLIFIED VERSION OF THE HOPKINS-STRYKER FORMULA

From ED LETHERT, 3-time graduate from Minneapolis.

$$\Delta 4' = [L \text{ sens } (1w\text{-}4') + 10 \log Q] - 23$$

This is a simplified version of the Hopkins-Stryker formula that gives the $\Delta 4'$ value for the calculating contractor who prefers to use the formula but does not have a programmable calculator. It's just a simplification that saves some time and it is surprisingly accurate. I suppose it could also be used to find D_S as this is usually a short distance. You will note that it ignores the reverberant field which has very little effect on the direct sound at short distances from the source.



Sent in by NICK METAL, Vancouver, B.C.

COMMUNITY LIGHT AND SOUND LOUDSPEAKER MEASUREMENT DATA

The long awaited Community Light and Sound loudspeaker measurement data reached our hands during the Boston class. (CHUCK AUGUSTOWSKI of Brandy Brook Audio in Derby, CT brought us two copies.)

It is an exceptionally useful document and belongs in the library of *anyone* attempting to design sound systems. We believe the data is accurate, and very helpful comparisons between competitive devices are presented.

The data should not be used, for example, in calculating the efficiency of units in the absolute sense but relative comparisons between units are valid. Since the sensitivity measurements are made 4' from the front of the horn rather than the driver diaphragm (so we are told) and the crest factor of the bandlimited pink noise signal is not stated, absolute efficiency figures can not safely be calculated. However, the difference in efficiencies of the different horn and driver combinations is easy to do.

			160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0 KHz
HORN Emilar EH800 (90° radial)	COVERAGE ANGLE ° HORIZ.	-6dB						320	80	60	55	60	100	100	120	140	120	100	100	100	90	90	70
		-12dB							120	100	80	170	140	170	180	200	160	140	130	120	110	115	90
		-20dB									220			220			260	240	180	140	140	150	140
DRIVER Emilar EA175-16	VERT.	-6dB						80	140	140	140	120	140	160	180	200	80	60	50	50	30	25	25
SPL		-12dB										200	220	220	240	230	190	90	85	80	60	40	40
1W Pink Noise		-20dB											280	300	310		260	220	140	160	140	110	100
500Hz-16KHz	RESPONSE		IN dB					-15	-7	-4	0	0	-1	-3	-3	-5	-1	-1	-4	-1	-4	-10	-16
99.07 db at 4'	Q							4.2	4.5	5.4	5.3	5.6	4.5	4.0	2.6	2.1	4.7	8.3	5.7	5.8	9.7	12.4	22.8
100.78 db at 1M	D_I							6.2	6.6	7.4	7.2	7.5	6.6	6.0	4.1	3.2	6.7	9.2	7.5	7.7	9.9	10.9	13.6
HORN Community RH90	COVERAGE ANGLE ° HORIZ.	-6dB					70	70	60	90	90	80	85	90	80	90	90	90	90	90	90	85	80
		-12dB					120	120	120	140	120	120	120	120	110	120	110	110	110	105	110	100	95
		-20dB								220	180	180	180	240	170	170	170	150	150	160	140	130	110
DRIVER Emilar EA175-16	VERT.	-6dB					140	160	100	120	100	80	60	60	55	60	60	55	40	40	35	35	30
SPL		-12dB					280	280	200	180	160	140	140	160	120	100	100	90	70	70	60	50	80
1W Pink Noise		-20dB									320	280	240	220	220	220	180	160	120	120	100	80	50
350Hz-16KHz	RESPONSE		IN dB				-11	-4	-3	0	0	+1	+1	0	+2	+2	+1	+1	-2	-1	-3	-8	-14
101.54 db at 4'	Q						5.5	4.8	6.9	6.1	8.1	8.2	9.1	7.1	11.5	8.6	9.9	9.4	11.2	9.9	10.9	14.2	20.6
103.25 db at 1M	D_I						7.2	6.8	8.9	7.8	9.1	9.2	9.6	8.5	10.6	9.3	10.0	9.7	10.5	10.0	10.4	11.5	13.1

$L_S - 10 \log Q$ = relative effic. Where L_S is the 4'/w sensi.

Thus, the Emilar driver on the Community Light & Sound RH 90 horn has a relative efficiency at 2000 Hz of

$$101.54 - 10 \log 7.1 = 93.03$$

And on its own Emilar EH 800 horn (assuming near equal horn lengths)

$$99.07 - 10 \log 4.0 = 93.05$$

This shows that neither horn restricts this driver's efficiency.

This excellent reference is worth many many times the modest $3 that Community asks for it. I'm sure that in the future they will provide the necessary information to allow absolute efficiency calculations by adding the pink noise crest factor of their generator sine waves and provide the distance correction back to the driver for the sensitivity measurements of all the horns tested.

THE HOT SPOT T.M.

During the Kansas City class, in addition to the wettest welcome ever, we tested a new stage monitor called The Hot Spot T.M. It is capable of 115 dB-SPL at 1M with a 100 watt input. Mounted on a standard microphone stand, it is able to be adjusted for best location to the rear of a microphone and at the same time minimum D_2.

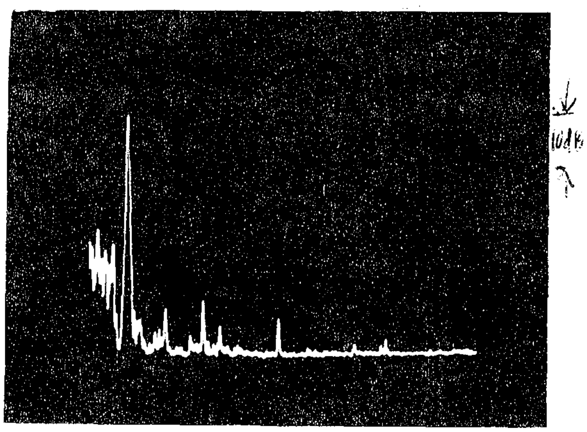


Figure 1

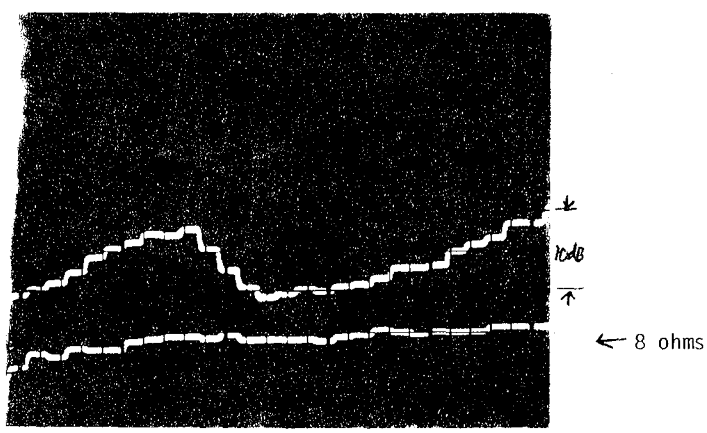


Figure 2

The distortion measurements (Fig 1) were made at high levels and show very low distortion components. The impedance curve is unusual (Fig 2) for two 5" loudspeakers and indicates careful enclosure design. The Hot Spot T.M. is manufactured by Galaxy Audio, 1417 E. Central, Wichita, KS 67214. (316) 263-2852.

IVIE IE-10A HANDHELD AUDIO SPECTRUM ANALYZER

We have now had a chance to evaluate the Ivie IE-10A handheld audio spectrum analyzer in some three classes. We have found it to be inherently an accurate useful instrument. In the Kansas City class there were three handheld RTAs. And in the Boston class we were able to compare it with the other handheld octave band analyzer on the market, much to the credit of the Ivie. The IE-10A calibration is quite accurate.

Our initial unit developed an intermittent nicad battery and after that problem was corrected, we found the unit easy to use and reliable. (We haven't talked to anyone who owns an Ivie that isn't enthusiastic about it.) The internal-external switch is too flimsy and broke after only slight useage. We believe Ivie can and will rapidly correct these minor mechanical problems.) The Ivie also operates as a conventional SLM at the throw of a switch.

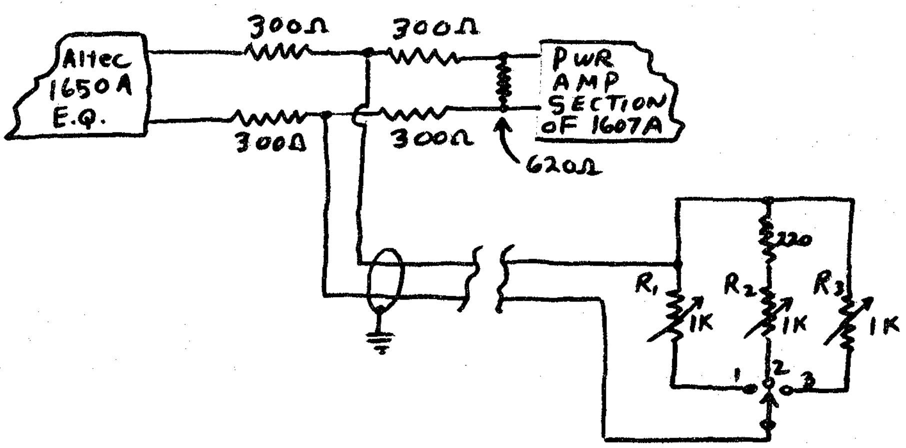
After using this instrument and experiencing the freedom of measurement a battery handheld unit allows, we eagerly await a 1/3-octave version with an external connection (calibrated) for use with a precision GR 1/2" electret microphone and a simultaneous output to a level recorder (for permanent records). The present unit's accurate calibration along with their really useful IE-20A pink noise generator (See Newsletter Vol 4, no. 4) makes tests of the type described by Andy De Ganahl of DisneyWorld both easy and accurate.

A SIMPLE REMOTE VOLUME CONTROL

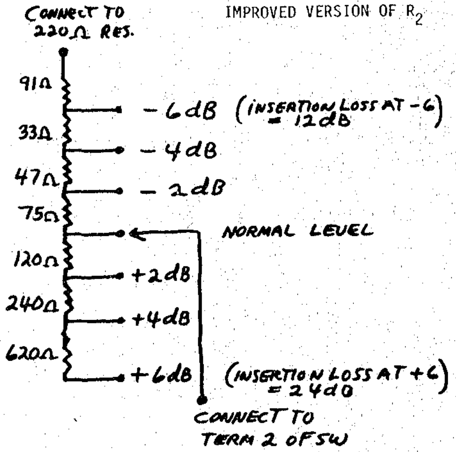
From ED LETHERT, Independent consultant, Minneapolis: This one is really neat. We cooked up a circuit for a church. Their problem was that one pastor spoke with a very soft voice and the other pastor was very loud. Both pastors participated in the same services and shared the same microphones. Also guest speakers and lay readers used these mics. At first I envisioned a rack full of equipment and then came a beautiful revelation. What if I installed a balanced "H" pad in the link between the equalizer and the power amp, remoting out only the shunt resistor over *one* twisted shielded pair.

We tried it and it worked!

Then what if we added other shunt resistors and a selector switch? Presets and presets and more presets were available over the *one* twisted shielded pair. It is now installed and working beautifully. They love it. I love it.



IMPROVED VERSION OF R_2



R_1 R_2, R_3 are 1K linear pots.

R_2 and R_3 are set and locked after adjusting the levels for the two pastors. R_2 is available for variable requirements.

The 220Ω resistor in series with R_2 keeps the system from being completely shut off and allows an adjustment range of roughly 6.5 dB (-12 dB to -18.5 dB). By decreasing the size of the series (220Ω) resistor the range can be increased.

The formulas are

$$\text{Loss dB} = \left[20 \log \left(\frac{\dfrac{R_1 \times 1200}{R_1 + 1200}}{\dfrac{R_1 \times 1200}{R_1 + 1200} + 600} \right) \right] - 6$$

(I know that dB loss here should be a positive number and my program does change the sign.)

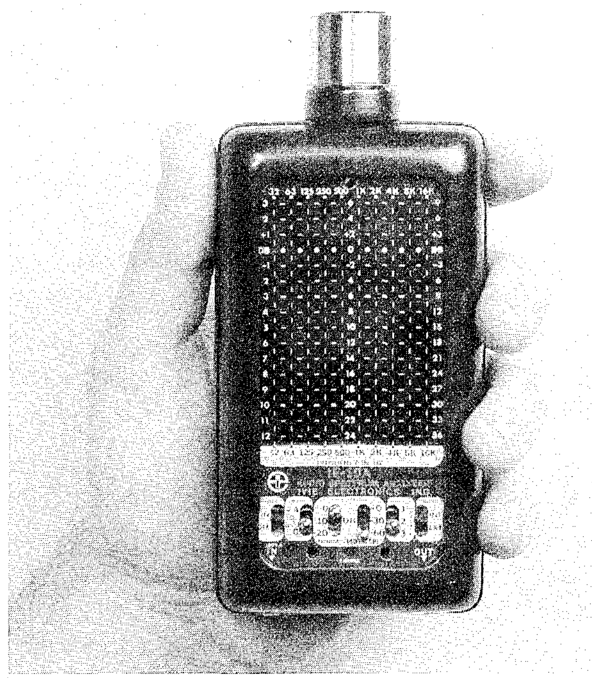
$$R_T = \left[\frac{600}{\left[10^{\frac{(dB-6)}{20}} \right] - 1} \right]$$

Note : The purist might not like the lack of a perfect impedance match here, but it has caused us no problem as far as we have been able to tell.

$$R_1 = \frac{1200 \times R_T}{1200 - R_T}$$

TAPE DECK ALIGNMENT WITH IVIE RTA

ANDY De GANAHL of DisneyWorld (and a two-time graduate) was one of the first users of the Ivie RTA so we asked him to give us a report, which he did several months ago. One of the uses for the Ivie that he mentioned was tape deck alignment, so we asked if he would write it up and share it in the Newsletter:



One of the most tedious chores of the amateur recordist or professional recording engineer is the periodic alignment of tape recording equipment to a new or different type of tape. Many home recordists with consumer machines are content with switching brands of tape and playing with the two or three position Bias and Record EQ switches (if any) until the deck "sounds o.k.", never realizing that in most cases their machines are capable of much better performance by adjusting the internal continuously variable Bias and Record EQ controls. However, this calibration requires some fairly sophisticated equipment including a variable sinewave generator and an accurate and sensitive AC voltmeter as well as a good alignment tape for playback calibration and plenty of available (and perhaps expensive) time. Most of this time has been spent in calibrating the Bias and Record EQ controls since a series of discrete frequencies usually had to be recorded to get the overall frequency response of the machine. With the introduction of the Ivie IE 10A octave band real time analyzer and companion IE 20A pink noise generator, much of the time and effort of this process can be eliminated.

As one of the first owners of the Ivie equipment, I have used the package successfully in many "live" sound reinforcement applications as well as for tape machine alignment. Since the RTA can display the energy in octave bands of an audio signal at any time and the PNG creates a signal with equal energy in each octave band, the combination can measure the frequency response of almost any device almost instantly. By inputting the "flat" pink noise to the device and reading the output of the device with the RTA one can instantly see what effect any control of the device has on any portion of the audio spectrum. Also, since the inputs and outputs of the Ivie units are RCA phono jacks, interconnections to a consumer type deck can be made quickly and easily.

Playback equalization can be checked with either a pink noise or a discrete tone alignment tape. The detailed Ivie IE 10A manual shows how tones between the octave center frequencies will read on the display. (Playback EQ is factory set and is usually very accurate, however, if any large problems in playback response are found, the Ivie will quickly tell if adjustments of the Playback EQ control can solve the problem.)

Connect the output of the first channel to be checked into the RTA. Roll the alignment tape and set the output level controls and the pads on the RTA for a good display. If the alignment tape is pink noise, adjust the Playback EQ control until the RTA readout is as close to a straight line as possible. If the tape is tones, write down the level of each tone and with the aid of the EI 10A manual determine the adjustments needed for flat response. Repeat for each channel by changing the connection to the RTA.

Once the playback EQ is set , load the machine with the tape to be biased for and connect the sine wave generator to the input. The generator and input level control should be set to give roughly the following input levels depending on the speed of the machine; 15 ips-0 dB; 7½ ips - 10 dB; 3-3/4 ips - 20 dB; to avoid high frequency self-erasure.

The bias is usually set by recording a tone while monitoring off the playback head and adjusting the Bias control until the output reaches the maximum level and then increasing bias until the output drops by a certain number of dB. (Using the Ivie RTA in its SPL-C mode makes this measurement easy to read since it converts the display to a single line.) However, the frequency of the tone and the number of dB of overbias depends on what the manufacturer of the tape recommends. (On my TEAC 3340 using Ampex 456, I overbias two or three dB at 10kHz.)

If the bias information is not available, there is a rather unorthodox method that I have used at 15 ips and 7½ ips to find close to the correct bias point. (It seems to work for me, but if there are any factors I am unaware of please let me know.) Record a 10 Hz sine wave on the tape while listening to the playback and monitoring the output with the RTA. Set the Bias Control for the minimum of modulated tape hiss at the output. This will give a reasonable approximation of the correct bias point. Also listen to the input to the deck to confirm that the extra playback noise is a result of increased distortion due to incorrect biasing and not the distortion being put out by the sine wave generator or the tape deck's record electronics.

Now that the bias is set for the point of lowest distortion, the Record EQ control is used to restore flat frequency response. Connect the Ivie pink noise generator to the input of the first channel to be set, keeping the input level correct for the tape speed being used. Record the pink noise and monitor the output with the RTA. Adjust the Record EQ control for the flattest response. Repeat for each channel.

The only problem that might occur at this point is if the latitude of the deck's Record EQ control is not wide enough to correct for the change in bias point. This is more likely to occur on older decks. In this case, a compromise must be found by setting the EQ at the extreme position closest to the desired response and adjusting the bias to restore flat response (increasing bias if the EQ could not attenuate the high frequencies enough, or decreasing bias if the EQ could not boost the high frequencies enough.) In either case, the instant readout of the RTA shows exactly what frequency response is coming off the tape at any time.

Once the user becomes familiar with the tape deck and the Ivie units a record-play alignment becomes a very fast procedure. I have done all four channels of my TEAC 3340 in less than twenty minutes. This is assuming all mechanical alignment has been done, tape tensions checked, heads and tape path cleaned etc.

Dr. John Hilliard is a Fellow of the AES *and* the ASA *and* the SMPTE *and* the IEEE. All professional groups recognize his pioneering efforts in audio and acoustical engineering, not only for its originality at the time but its fundamental permanence as the years pass.

While many of us have employed the concept of "plugging" individual cells of a multicellalr horn, Dr. Hilliard has taken the time and effort to identify the design values for a given result. We are very pleased to be the avenue of communicating this excellent data.

ACOUSTIC ATTENUATOR FOR MULTICELLULAR TYPE HORNS

by

Dr. John Hilliard

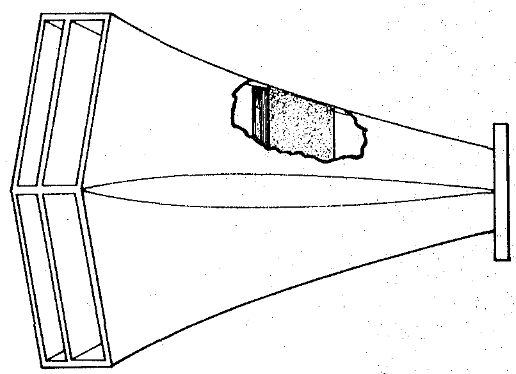
There are many situations in a sound re-enforcing system where it is desirable to change the distribution pattern of multicell horns by eliminating or reducing the transmission out of some of the cells.

Plugging a cell causes reflection back into the throat of the horn with an undesirable interfering pattern. If material is placed in the cell which absorbs the sound without reflection, destructive interference is avoided. An ideal material for this purpose is lambs wool, fiberglass or similar material having a characteristic impedance equal to that of air.

When a wave strikes another medium, reflection occurs and the ratio of pressure to particle velocity and the pressure are in phase or P/V = Space or the characteristic impedance where Space (RHO) is the density of the air and "C" is the velocity of sound in air.

Fiberglass wedges having a weight of 4-6 lbs/cu.ft. are sufficiently firm that it can be cut and shaped to fit into the horn. The attenuation of any fiberglass material having a density of more than 1 lb/cu.ft. will have a loss of approximately 1 dB per inch of thickness. These wedges are easily prefabricated and can be wrapped with a very thin limp film no thicker than .001 inch or any suitable mesh cloth or screen.

RHO-C ATTENUATION WEDGE



LAMINATED FIBERGLASS DUCT BOARD (4-6# DENSITY)
ADJUST LENGTH FOR DESIRED ATTENUATION
OPTIONAL - LOOSE FIBERGLASS IN .001 INCH THICK PLASTIC SACK

Since they have the characteristic impedance of air, I have called them "RHO" wedges. They can be used in many installations and avoid changing an existing horn to secure the desired distribution pattern. Other materials such as open cell polyurethane having the same density may be used. The same technique can be used to modify the distribution pattern of sectoral horns.

(ED'S NOTE: for a wedge that is 12" thick, you can calculate 3 dB + 2 log 12 = 5.16 dB of attenuation. The attenuation effect of fiberglass having this density (higher densities would cause unfavorable reflections) is initially linear (i.e. for the *first* 3" the attenuation is 1 dB/inch) and for thicknesses greater than 3", it becomes a log function (2 log total depth) added to the initial 3 dB.)

DEFLECTION OF THE EARDRUM AT VARIOUS SOUND LEVELS

If we make the assumption that the eardrum displacement is the same as that of the molecule striking it we can write:

$$D_{in} = \frac{3 \times 10^{-7}\left(10^{\frac{dB\text{-}SPL}{20}}\right)}{f} \quad \text{or} \quad D_{cm} = 3.9 \times 10^{-3}\left(0.0002 \times 10^{\frac{dB\text{-}SPL}{20}}\right)$$

Where D_{in} is the displacement in inches (the RMS amplitude) of the air molecule

D_{cm} is the displacement in centimeters

f is the frequency in Hz

dB-SPL is the sound level in dB referred to 0.00002 N/M^2

An example:

For a tone at 1,000 Hz at a level of 74 dB

$$D_{in} = \frac{3 \times 10^{-7}\left(10^{\frac{74}{20}}\right)}{1{,}000} = 0.0000015''$$

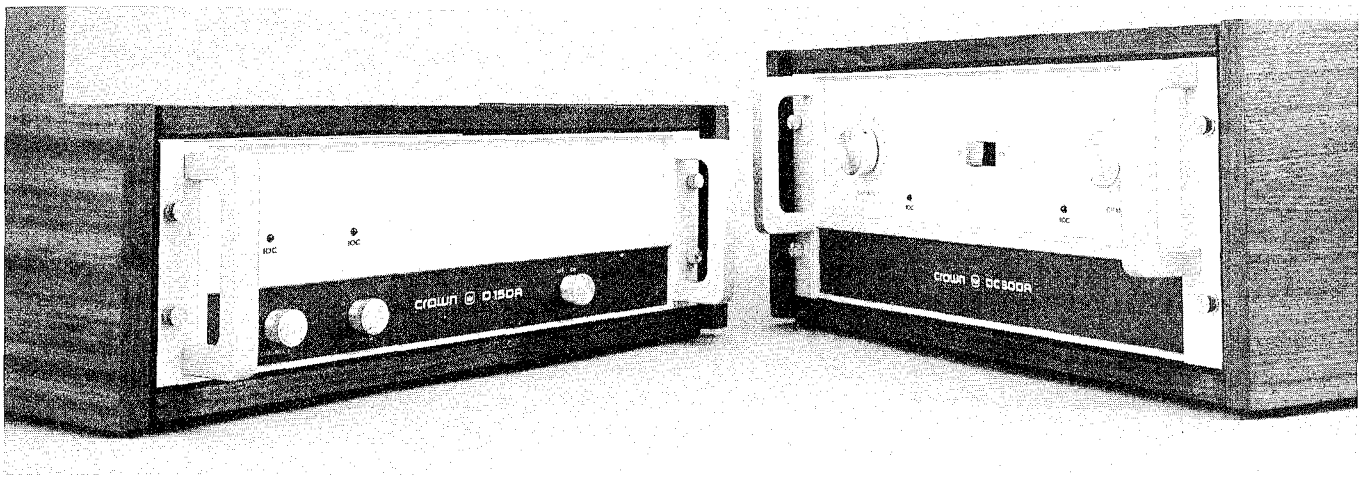
A displacement of approximately $\frac{1}{1{,}000{,}000}$ of an inch.

I O C - THE MUSIC DISTORTION INDICATOR FROM CROWN

Technical Bulletin from Crown:

The input-Output Comparator system now incorporated into the D-150A and DC-300A represents an important departure from traditional clipping/overload indicators. The circuit uses a simple principle to report to the user some very important information.

Inherent in the design of Crown amplifiers is the presence of a correction signal anytime the feed-back loop indicates a difference between the input and output of the amplifier. The input IC acts as an input and output comparator on a continuing basis. When any kind of overload or nonlinear behaviour appears, the IC generates a correction signal in order to match the output to the input again. The Crown IOC network monitors this correction signal and lights the indicator LED when a correction signal indicating any kind of overload is present.



The advantage over the traditional clip indicator becomes apparent when the action of such a circuit is considered. Sensitive only to output voltage, the more common LED clip indicator can not respond to protection circuit activation (caused by either signal or load problems) or slew induced distortion (commonly called Transient Intermodulation Distortion or TIM), and may not even report amplifier clipping if the overload is brief. The IOC, by contrast, reports *any and all* forms of amplifier overload or non-linear behavior, including SID, protection circuit activation, and all amplifier clipping no matter how brief. The circuit sensitivity is extremely high, as the front panel display is activated *before* amplifier distortion specifications (.05% THD, IMD) are reached.

For the first time, without complex laboratory equipment, the user can "see" and avoid distortion caused by any type of amplifier nonlinearity. And, most important, this can be accomplished under actual listening conditions.

Because of the extremely useful information that IOC provides - and in keeping with Crown's policy of avoiding continual model changes and planned obsolescence, existing owners of DC-300A's and D-150A's may have their units updated for $60.00. Units must be returned to the factory as the cost includes a laboratory check out after IOC is installed. Suggested retail prices are being increased nominally ($50.00) for the new units with IOC.

COMMENTS BY A "DISCO" EXPERT

ALEX ROSNER of Rosner Custom Sound in Long Island City (and a many time graduate) has designed, installed and backed up over 250 audio systems in Disco clubs during the past 12 years. Currently Alex is installing a $50,000 (sound system only) job. Some key points that Alex shared with the New York 1977 class were:

1. Reliability MUST be achieved. A "down" system means the loss of that evening's revenue for the club owner. Therefore, redundancy is a big part of system planning. Alex suggested that losing up to 1/2 the system should be tolerable for one evening.

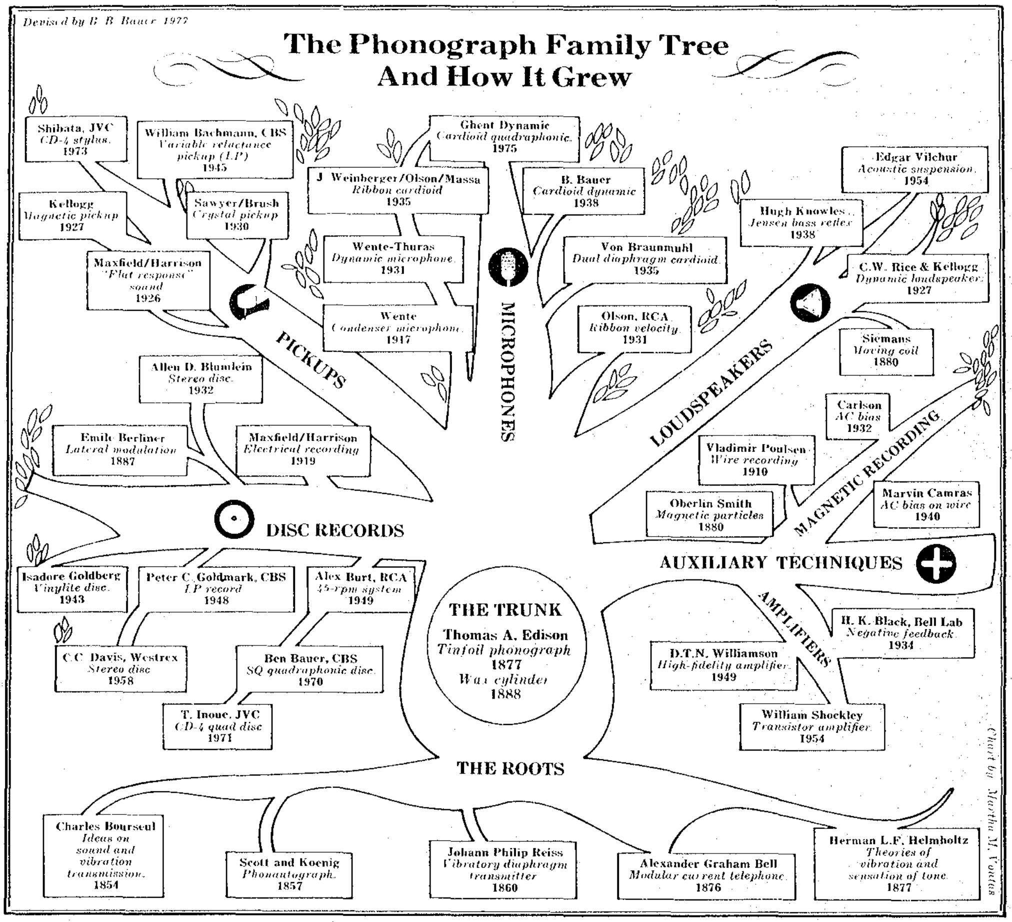
2. Equipment must be high quality, stable, and capable of having all varieties of liquids poured on, in, and over without shorting, etc. The mixers used are the weakest link at present.

3. Physiological Data: Alex has found that the preferred "beat" in Discos is the heart beat rate. He has determined that the bass units MUST be on the floor as apparently enough bass is required to stimulate the thighs of the dancers. The *HANDBOOK OF NOISE CONTROL* edited by Cyril Harris, Chapter 11 deals with the "Effects of Vibration on Man" and Figures 11 and 12 covers the mechanical impedance of the surface of the thigh. Two frequencies would appear to be important to the massage of intimate organs, namely the regions around 14 Hz and 40 Hz.

4. Alex uses the new McIntosh amplifiers solely because they automatically limit the input signal whenever the output signal reaches a predetermined value.

5. Get the money FIRST, then do the job. In his 250+ jobs, Alex has lost only one job by asking for the money up front. As a side remark, the contractor who did get the job never collected.

Our thanks to Alex Rosner for his willingness to share hard earned knowledge.



Ben Bauer, a prolific inventor-scientist, devised the "family tree" shown above. Under "Auxiliary Techniques" I would have added Dr. Boner and Don Davis, Room Tuning. And I would have placed Paul Klipsch, folded corner horn - 1941 - in place of Edgar Vilchur - acoustic suspension 1954. I believe that Lord Rayleigh belongs among the Roots. (Written by Carolyn Davis)

MAC 77

We have been pleased to hear from many graduates who attended the Midwest Acoustics Conference in Chicago that they did indeed find it useful, informative, and entertaining. TIM KELLEY's letter (Sound Dispersion Services, Balsam Lake, WI - Nashville 1976 class) is typical.

> Just a note to tell you how much I enjoyed MAC 77. It was very much worth the time and effort to get there. There were a number of good points brought out that I think would make good Newsletter material. For example, Stan Miller and his reference to splitting mic lines for mixing and recording; David Klepper and his reference to the light method for adjusting arrays.

PAUL KLIPSCH'S TALK ON DISCOTHEQUES

STUDIO SOUND from England wrote in great detail about the Los Angeles AES Convention and ended the article with "the session attracting most attendance was Paul Klipsch's talk on special speakers for discotheques, which had 650 eager listeners."

If you read that and if you were like us, you wondered how you could have missed a paper by O. Gadfly Hertz that 650 eager listeners did not miss. We wrote Paul Klipsch about it. He said he was sorry he missed it too. He wrote, "I wish I could have heard it, although I would probably have started an argument with the author." So, quit worrying if you missed it. So did Paul Klipsch.

AUDIO CYCLOPEDIA

We found that we didn't have time to be Senior Editor for the revision of *AUDIO CYCLOPEDIA* and recommended GLEN BALLOU to Howard W. Sams as editor. Glen writes:

The *AUDIO CYCLOPEDIA* revision is going high speed forward. As you know, much of it is wrong and outdated, as well. To make a worthwhile reference book, it must have inputs from as many people as possible and the best place to get this is from Syn-Aud-Con graduates. So please send in your ideas, thoughts and corrections. Credit will be given for major ideas and corrections.

PERTINENT COMMENTS ON MOTION PICTURE SOUND

Recently while engaged in disagreeing politely with an author's article in a technical journal I found myself in extended correspondence with the author I had criticized. Wishing to be very sure that my own beliefs were not masking a middle position on the subject, Carolyn sent the correspondence to our distinguished graduate, TED C. UZZLE of Boston, Mass. asking for his comments. When they arrived they transcended both mine and the criticized author's work so thoroughly that we have obtained Ted Uzzle's permission to reprint his letter. He goes into the matter with such care that the previous correspondence (well described in his first paragraph) is neither necessary or worth reading.

While the letter proves it, we will mention in passing that Ted Uzzle earns his livelihood as a motion picture sound consultant. Ted is also the editor for the sections on motion pictures in the *AUDIO CYCLOPEDIA*.

Dear Carolyn:

The exchange between Don and Mr. X on which you have permitted me to eavesdrop, is a dialogue of polite mutual condescension worthy of Jane Austen.

Discussion of the 1ms integration period is rather beside the point because Mr. X does not use it in any derivations in his paper. I would have to agree with Don that a delayed loudspeaker will continue to perturb localization away from the original source when the delay is rather longer than 1ms, perhaps even up to 10ms. We know from experience with stereo at home that localization does not jump to one speaker only, even when we move considerably off axis and when we are playing monaural material.

As to theatrical stereophony, I am familiar with many stereo installations, including wide auditoria. These are usually characterized by seats with good sound and seats with poor sound. Still, I have never experienced the precedence effect between loudspeakers in a theatre sound situation, nor do I know anyone who has. I doubt if Mr. X has. Where only the near loudspeaker is heard at a front corner of a theatre, we are invariably able to correct the problem by turning the far loudspeaker(s), which demonstrates that it's a coverage angle problem and nothing more.

Mr. X is dead on the money when he remarks in passing that theatre stereo is more sound effect than sound, but the cause has nothing to do with delay, precedence, nor even with acoustics at all. The best exposition of this distinction I've ever read is in the transcribed discussion between William Snow and John Frayne in J-SMPTE 61:5 p 587 (Nov 1953).

"Pop-out time" is much more complex than the 1ms or 10ms period. By pop-out time I mean the minimum delay required for the delayed signal to pop out from the psychoacoustical shadow (as it were) and make itself heard as a discrete echo. Mankovsky, in Acoustics of Studios and Auditoria, pp 14-15, quotes Haas' 1951 paper in Acustica to the effect that pop-out time varies with speech tempo, the absolute and relative intensities of the signals, the frequency spectra, reverberation time, and last and perhaps least, individual perceptual anomalies. He cites pop-out times from 40ms to 250ms.

Kuttruff, in section VII.3 (Annoyance Due to Reflections) of his book Room Acoustics, goes into these variations in some detail. If you will inspect fig VII.5 at p 174, you will find effects of syllable rate, with 50% of listeners overall finding intelligibility degraded with a delay of 68ms. Mankovsky would seem to support Mr. X's 65ms figure in fig 2.36 at p 75 of his book, in which he shows the first 62.5 ms of reverberative decay as "helpful to the signal" and the remainder as "noise". I would surmise that this entire section, beginning at p 73 and dealing with the "liveliness coefficient" was near Mr. X's hand as he wrote the SMPTE paper.

It would seem that both Don and Mr. X have failed to specify completely the conditions under which they expect 65ms to pop or not to pop.

We're talking about review rooms and cinemas of current design; I'd have to go along with Don that 65ms is just too long. We should draw the line somewhat lower, around 45 or 50ms. This presupposes all those conditions that actually exist in the cinema theatre and Don's delay demonstrations. The fact that we can accept longer delays when the band strikes up or during slow speech is essentially immaterial, since we expect nearly 100% comprehension in both the theatre and the lecture hall.

To understand the significance Mr. X attaches to a 65ms delay, we should consider these remarks from his article in dB Magazine (10:2 p 25 Feb 1976):

> I have always been curious about the ubiquitous factor of sixteen that we find as the minimum frequency discernible as a tone, the minimum flickerless frame rate for motion pictures, and (as a reciprocal) the maximum fusion time for echoes.

As you can see, Mr. X has something I don't have: A Grand Theory. Or at least a Significant Clue. Personally, whenever I fancy I know something about psychoacoustics I take Green's and Swet's Signal Detection Theory and Psychophysics down from the shelf. I'm now able to get as far as the receiver operating characteristic before collapsing into neurasthenia.

Sixteen may induce epileptic seizures; it may even cure the clap; but it's too damn slow to perceive as a tone or smoothly moving picture.

Edison's first cameras and peepshow viewers used more than 30 frames per second: he needed that many, he found, to reduce flicker to tolerable levels and create the illusion of smooth motion. The consumption of film and the wear on the mechanism were so great, however, that he decided to use 16 frames per second, each frame viewed twice, for a field frequency of 32Hz.

(With the years this increased, and in 1924 Western Electric engineers with recording tachometers found a 24 frame per second, 48Hz field rate, to be the average for silent pictures shown in Chicago theatres. At the Lake Placid SMPE convention Western Electric prevailed with this figure as a standard for the talkies over proposals by GE (RCA), Fox-Case, and others. Thus, the conventional wisdom that "silent was 16 frames but sound pictures were speeded up for fidelity" is historically wrong on both counts.)

....continued next page

COMMENTS ON MOTION PICTURE SOUND BY TED UZZLE, CONTINUED

(Even the modern theatrical motion picture field frequency of 48Hz is too slow to prevent flicker disturbance in some high-illumination applications: Todd-AO used a frame rate of 30 for a field frequency of 60Hz, and small review rooms often project each of 24 frames per second three times each, for a field frequency of 72Hz. There's a lot of variation according to illumination intensity, visual field subtense, and other factors, but 16Hz isn't even in the ballpark.)

I have pondered the references in Music, Acoustics, and Architecture, and don't really see that they apply either way. One aspect of Beranek's desideratum of early reflections (c. 20ms) in the concert hall may have to do with the phenomenon by which early echoes fool the ear and prevent later ones - well beyond 65ms - from popping out as discrete echoes. I believe this is what Kuttruff shows in fig VII.3 at p 170 of his book.

All this is immaterial. I don't really understand how the references to 1ms and 65ms came to occupy so much of the correspondence.

In his paper Mr. X made essentially the same point as "The Equalization Myth" by Alan Fierstein in the current number of dB Magazine. And he made essentially the same mistake. He offers formulas by which, given the steady state response measured in the reverberant field (and a lot of other information), one can calculate the direct+early spectrum and the reverberant spectrum separately. He then proceeds to apply this to situations where there is no true critical distance and no true reverberant field. As Cecil Cable wrote, joining Don in a reply to George Augspurger (who made this same conceptual error), in dead rooms there will be many points, at varying distances, where time delay spectrometry will reveal equal quanta of direct and reflected energy, but these will be caused by reflection anomalies, speaker lobe nulls, and other phenomena having nothing to do with the statistical acoustics (paraphrased from J-AES 24:3 p 190, April 1976).

When Mr. X introduces the pressure-squared equation, #7 in his paper, he presupposes a condition that does not exist in mixing rooms or review theatres of the volumes and reverberation times he posits. Don is absolutely correct in this matter, maybe even more so than he realizes.

What about the public cinemas? There's something interesting going on here, and we ought to look at some contemporary theatre design criteria. (Bet you knew I'd get around to this sooner or later!)

Between 1970 and 1976 the number of public cinemas in the United States increased from 9,000 to something over 15,000. For a variety of reasons, the vast majority of this increase was in the form of multi-cinemas and "splits", existing theatres divided with a wall down the middle, across the proscenium, or from the lip of the balcony upwards. The old Loew's Uptown Toronto was split into five mini-cinemas in the pattern that must be seen to be believed.

The motion-picture industry is responding to the same economic pressures that have shaped magazine publishing and radio broadcasting: the broad appeal is gone, and what's left are many fragmented special interest vehicles. The economically viable theatre of today has 600 seats or less and is in a building with several other theatres. The movie-going family splits up at the boxoffice and all the different folks get all their different strokes.

These theatres, by geometric necessity are long and narrow. In my talk to our local SMPTE and AES sections last year I presented a derivation by which theatre dimensions were expressed as a function of the number of seats. Inspecting the results in the interval from 200 to 600 seats, I found that reverberation times of one second are impossible in these theatres even if there is no absorption except the audience. The practical limit of reverberation may be expressed by Rettinger's equation

$$RT_{60max} = \frac{1}{\dfrac{18.2\ S_f}{V} + 0.1}$$

Where V = room volume

S_f = surface area of floor
English units

(Acoustic Design and Noise Control, 2d edn, p 97)

In any rectangular room the ratio V/S_f becomes simply the ceiling height. Using standard design criteria we can express both V and S_f in terms of n, the number of seats, whereupon n drops completely out of this equation and RT_{60max} takes the value of 0.85 seconds.

(I think Knudsen was the first to use this technique in print. I picked it up from Carbonel and Zuccoli, "Volume Per Seat and Variation of the Reverberation Time With the Size of the Audience", J-ASA 33:6 p 757, June 1961. I regard my application as simply a specific use of a common technique and have omitted the full derivation and criteria here for that reason)

Those of us who take reverberation timers into theatres on a regular basis find values closer to 0.4 and 0.5 seconds simply because there is absorption in addition to the audience. The largest theatre chain in this country (something like 600) has standardized on a wall treatment of perforated metal in front of fibreglass blankets with absorption coefficients reported by the Riverbank Labs to be:

125	250	500	1000	2000	4000
0.74	0.97	0.87	1.02	1.05	0.95

Now, Carolyn, just imagine putting a speaker with a midrange Q of 5 or more in the center of one end wall and pumping sound up a 400 seat mini-cinema 30 x 18 x 95 with an audience and that stuff on the walls. No way on this planet Earth is there going to be a true reverberant field.

What I am saying is that in the United States in 1977 we mix, dub, sell, and watch movies within critical distance almost entirely. There is no widespread auditing of motion picture sound from the reverberant field, and the reverb times are almost invariably below 0.85 seconds. When Vlahos writes

> Acoustic response measurements are made in the reverberant field. In most rooms all but the front quarter of the theatre is in the reverberant field. (J-SMPTE 84:1 p 7 Jan 1975)

I want to jump up and down and scream and rend my garments. It just ain't so. They don't build'em like that anymore.

....continued next page.....

COMMENTS ON MOTION PICTURE SOUND BY TED UZZLE, CONTINUED

Mind you, this is not to say that the proposed acoustic response standard is acceptable because the characteristics of review rooms and most public cinemas are more similar than has been supposed.

Let's look at some of the techniques and instrumentation that have been used in the literature and in proposed standards. We may omit the pioneering work of Durst, Mason and Moir, Mueller, and others who led the way in the 1930's and '40's; let's pick it up eight years ago.

In an ISO draft proposal (J-SMPTE 78:12 p 1045 Dec 1969) we find a 1/3 octave filtered pink noise generator supplying the signal and measurements made with a handheld SLM with 1" microphone with random incidence corrector. That's the proposed standard: a 1" microphone with random incidence corrector.

At the same time Ljungberg (J-SMPTE 78:12 pp 1046 - 1053 Dec 1969) was using 1/3 octave filtered pink noise recorded on ¼" tape and played back in the field on a Nagra. He took measurements with a SLM with no wave analysis.

Rasmussen (J-SMPTE 78:12 pp 1058 - 1063 Dec 1969) used a 1/3 octave filtered pink noise optical sound test film, a calibrated measurement microphone and a strip chart recorder.

Vlahos (J-SMPTE 84:1 pp 3 - 7 Jan 1975) apparently used an optical sound test film with octave bands of pink noise successively shifted up 1/3 octave at a time, perhaps to create a statistical 3-bin averaging window. His measurements were the corrected average of several readings with a handheld SLM.

Rasmussen, writing a second paper (J-SMPTE 85:3 pp 164 - 169 March 1976) used the Nordisk Film/TV Union optical sound test film with octave filtered pink noise at 9 dB below 100% modulation. Readings were made with handheld SLM with integral octave band analyzer. It took 6½ years for the perception to break into print that band-limited pink noise and a SLM without analyzer aren't a useful combination. What is truly astonishing, even stupifying, is that over the entire 6½ years real time analyzers were available commercially from several manufacturers, at steadily tumbling prices. And, now that Rasmussen finally mentions real time analysis in a footnote, leading acousticians are turning from it to time delay spectrometry!

The techniques being used in the development of an acoustic response standard for cinema theatres are antiquities in a rapidly changing, rapidly developing field. You can see that I agree completely with Don that this standard is in deep, deep trouble, and is more likely to retard real progress than to advance it. "Edit out the rhetoric"? The more specific, the more purple!

We need to know much more about spectral and temporal behavior of sound in heavily damped rooms.

We need to know much more about cascading similar, short reverb times in the studio and the theatre (as in Mankovsky, pp 47 - 52, and Hill, "Combined Reverberation Time of Electrically Coupled Rooms," J-ASA 5:1 p 63 July 1932).

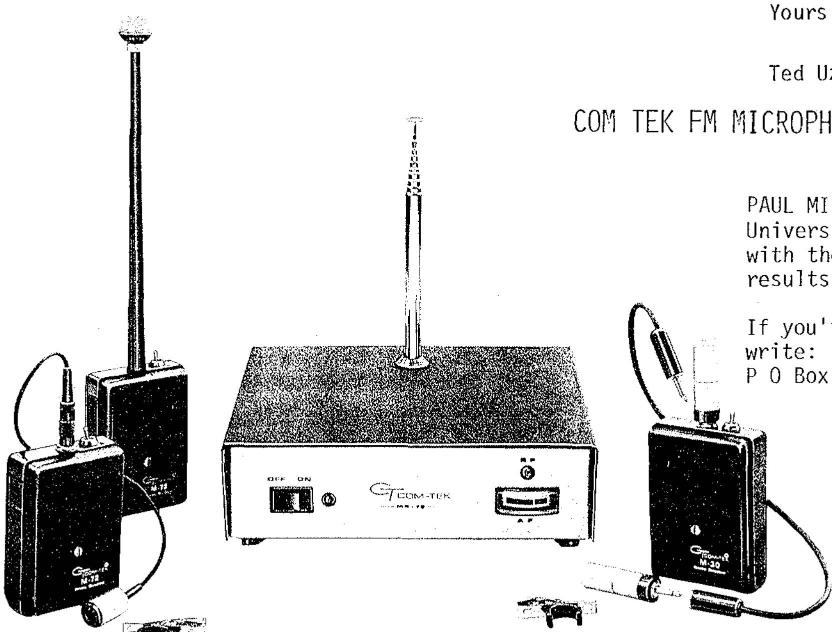
We need to know much more about the psychophysics of reproduced image and sound (as in Maxfield, "Some Physical Factors Affecting the Illusion in Sound Motion Pictures," J-ASA 3:1 pp 69 - 80 July 1931).

I don't know if there's any single statement or derivation in Mr X's paper I could point to and say, unequivocally, "That's wrong!" The man is no dummy, and for that very reason I wish he were writing papers that cut to the heart of the problem, rather than deriving formulas no practitioners in the field will or ought to use.

Yours very truly,

Ted Uzzle

COM TEK FM MICROPHONE



PAUL MINER, Electronics Department at Utah State University sent us info on the Com Tek Fm Microphone with the comment: "I've used these with excellent results indoors and out."

If you're interested in more information you can write: Communications Technology, 375 W. Lemel Circle, P O Box 15969, Salt Lake City, Utah 84115

NEEDED - RANDOM ACCESS TAPE DECK

TOM FABYANSKI (2-time New York graduate) needs a random access tape deck. Tom would like to be able to pick one out of 24 pre-recorded messages for use in a display. The decks have to be simple, semi-reliable, and inexpensive. He needs 3,000 to 5,000 units.

If you have any information to help Tom, contact him at R. J. Martin Co., 321 Commercial Ave, Palisades Park, New Jersey 07650. Phone 201/592-0952. Let us know in Tustin, also.

MODIFICATIONS OF THE SOUND SYSTEM DESIGN PROGRAM FOR THE HP 97

GARRY MARGOLIS, applications engineer for the professional division of JBL, has provided some very effective embellishments for our H.P. 97 sound system design program. The letter below indicates how some of the better "Wig Bubbles" are communicated.

Dear Don:

Many thanks for making the Sound System Design program available to us HP67/97 owners. Using your work sure beats having to write such a program myself!

I have modified the program slightly, and the modifications might be of interest to you. To wit:

1. In order to take advantage of the automatic card read feature available using the Pause, I have added this routine after step 110

```
LBL E
PAUSE
GTO E
```

With this routine inserted, the calculator will blink the Critical Distance figure arrived at until the second card is inserted in the slot, at which time the card will be pulled through. When the second side of the card is read, it will automatically initialize itself, and pressing "A" is not required.

Further, while the calculator is printing out $\%AL_{cons}$, Max RT_{60} and D_c, the second card can be inserted in the slot and will be called in automatically by the PAUSE instruction.

If it is desired to rerun Card 1, pressing R/S if the program is not already stopped, and "A", will initialize the card.

2. If you wish to rerun Card 2 and you have reached the end of Card 2, the way the program is written requires simply the pressing of "A", but if the program has not run its course, it is necessary to exchange primary and secondary registers before pressing "A". To avoid confusion, I have deleted instruction 104 on Card 2. In each case when the return to the start of the card is desired, it is then necessary to exchange registers and press "A".

3. Since Card 2 is an extension of Card 1 and not a separate program, I have changed the prompt step numbers by adding 6 to each of them. Step 009 ("1") has become "7", etc. This avoids confusion as to which card is working.

4. Those of us who have HP 67's have to wait five seconds after we enter each piece of data while the 67 pauses instead of prints.

To speed things up and avoid unnecessary waits, I have deleted the following steps:

Card 1:
010 PRTX
014 PRTX
018 DSP3
019 PRTX
020 DSP2
024 PRTX
045 PRTX
069 PRTX
173 PRTX

Card 2:
012 PRTX
018 PRTX
024 PRTX
030 PRTX
054 PRTX
053 PRTX
077 PRTX
095 PRTX
105 SPC
106 SPC
107 SPC
108 SPC
109 SPC

In each of the above modifications, I have used our original step numbers for reference. Of course, each time something is added or deleted, the step numbers will change.

Thanks for your metric modifications. Once again, you've saved me some work. Permit me two comments:

1. When changing 641.81 to 200.00, it is not necessary to key in the decimal and the following two zeroes, since the calculator automatically assumes that they are there unless otherwise instructed.

2. Since loudspeaker sensitivity in countries using SI units is specified at 1 metre, the only substitution needed on card two is to change step 006 from 4 to 1. Then, of course, the sensitivity will have to be entered as a 1W, 1m figure, not 1W, 4ft.

In analyzing your program, I learned a tremendous amount about programming in general and the HP 67's capabilities in particular. I must congratulate you for some elegant solutions to difficult problems.

If I have anymore wig-bubbles, I'll let you know.

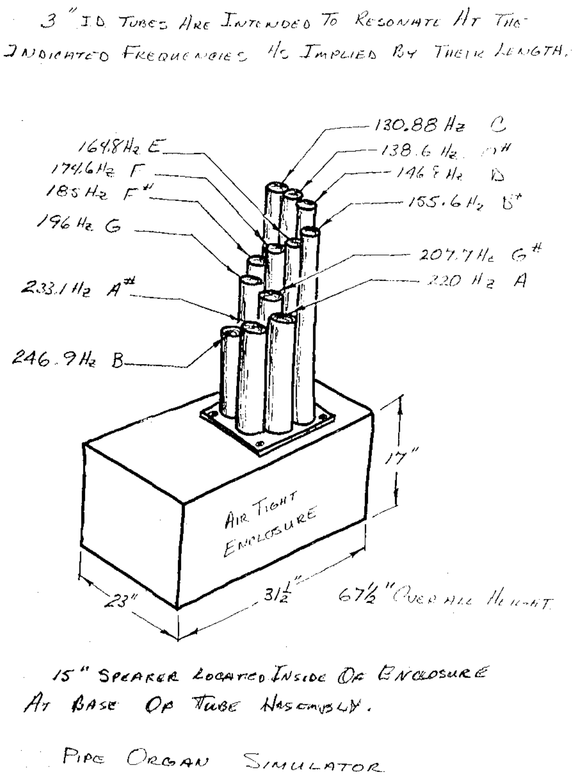
Best regards,

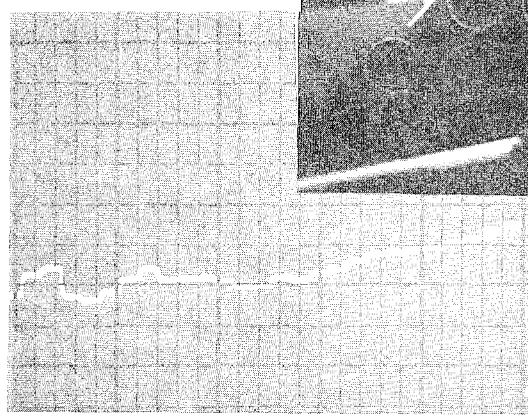
Garry Margolis

(EDITOR'S NOTE: We have reprinted the Sound System Design program incorporating the changes suggested by Garry)

"PIPE ORGAN" SPEAKER SYSTEM

We had a chance in the Minneapolis class to test a fascinating loudspeaker. See drawing. It performed surprisingly well and exhibited a remarkably smooth impedance curve for a woofer (see photo.





KARL KROPP wrote up how he came to build the system.

The "pipe organ" speaker system is found in the December 1971 ELECTRONICS WORLD magazine. The changes I have made I based on the fact that resonance depends on length not volume of the pipes. These pipes are approx. 3¼" I.D.. The tubes are directly coupled to a 15" speaker - silicone sealent was applied to any gaps. The cabinet is dimensioned as prescribed by Badmaieff and Davis's book, *How to Build Speaker Enclosures*. No damping material was used as the method described on page 77 fig 4-21 of the above book determined that none was needed. The cabinet is of 3/4" plywood. The tubes are cardboard carpet roll tubes.

DERIVATION OF THE LOGARITHMIC MULTIPLIER

By definition:

$$\mathrm{Log}_{10}\, 10 = 1 \text{ Bel}$$

By manipulation the power ratio of 1 Bel equals:

$$10^{(1 \text{ Bel})} = \frac{10}{1} \text{ (power ratio)}$$

By definition:

$$1 \text{ Bel} = 10 \text{ dB}, \quad \frac{10 \text{ dB}}{1 \text{ Bel}} = \frac{1 \text{ Bel}}{10 \text{ dB}} = 1$$

By dimensional analysis:

$$\frac{1\ \cancel{\text{dB}}}{1} \cdot \frac{1 \text{ Bel}}{10\ \cancel{\text{dB}}} = .1 \text{ Bel}$$

By manipulation the power ratio of .1 Bel equals:

$$10^{(.1 \text{ Bel})} = \sqrt[10]{10} = 1.26\ldots \text{(power ratio)}$$

or

$$\mathrm{Log}_{10}\, 10^{(1/10)} = .1 \text{ Bel} \quad \text{but } .1 \text{ Bel} = 1 \text{ dB}$$

By manipulation we can write:

$$\mathrm{Log}_{10}\, 10^{.1X}$$

or

$$X(\mathrm{Log}_{10}10^{1/10}) = 1 \text{ dB}$$

$$X = \frac{1 \text{ dB}}{\mathrm{Log}_{10}10^{1/10}} = 10$$

$$10(\mathrm{Log}_{10}10^{1/10}) = 1 \text{ dB}$$

FINDING THE INTERNAL VOLUME IN A LARGE SPACE THAT HAS COMPLEX GEOMETRY

A professional sound engineer wishing to design predictable articulation and acoustic gain into a sound system often finds that he needs, within reasonable accuracy, the internal volume of the church auditorium, arena, field house or other large reverberant enclosed space that desperately needs *professional* diagnosis before any more operations are performed on its acoustics and the owner's pocketbook. I have often wished for a push-button measurement of volume.

The acoustic consultant has normally applied the classic technique of laboriously calculating from available drawings or measuring, often with actual surveying tools, the internal volume (V) in ft^3 or M^3 and the boundary surface area (S) in ft^2 or M^2. Then by measuring the reverberation time for 60 dB of decay (RT_{60}) in seconds, he is able to calculate the optimum interface between the sound system and the acoustic environment.

While the professional sound engineer has the classic tools available and may, on occasion, use them, especially the RT_{60}, the advent of professional loudspeakers with their Q properly and accurately measured and specified allows a much quicker and just as accurate an approach to be considered.

<u>The Relation of Critical Distance, D_C, to Directivity factor, Q, and the Reverberation Time, RT_{60}</u>

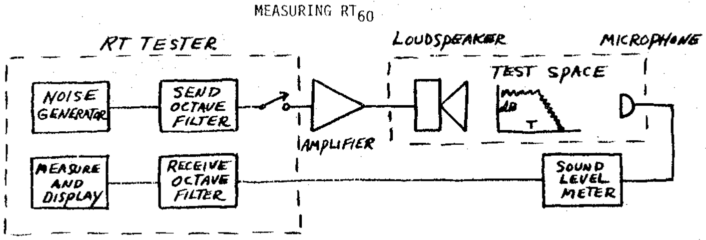
The equation

$$D_C = 0.03121\sqrt{\frac{QV}{RT_{60}}}$$

allows the interesting variation

$$V = \frac{(D_C)\ RT_{60}}{(.03121)^2 Q}$$

The measuring set up then, becomes that shown in Figures 1 and 2.



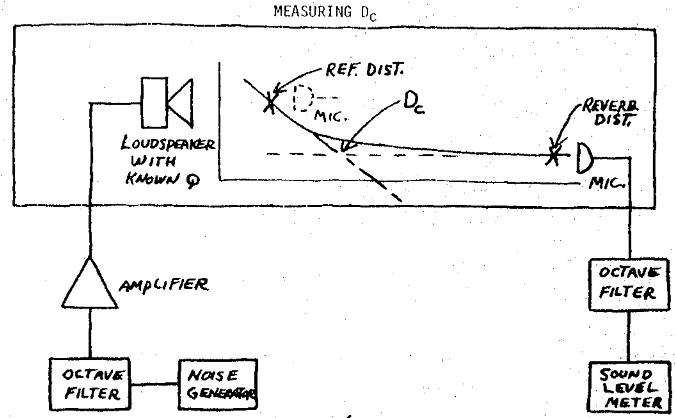
$$RT_{60} = \frac{.049V}{S\bar{a}}$$

$$D_c = .141\sqrt{Q S\bar{a}}$$

$$S\bar{a} = \frac{.049V}{RT_{60}}$$

$$D_c = .141\sqrt{\frac{Q \times .049V}{RT_{60}}} = .03121\sqrt{\frac{QV}{RT_{60}}}$$

Fig 1



$$D_c = \text{REF. DIST.} \times 10^{\left(\frac{dB_{REF.} - dB_{REVERB.}}{20}\right)}$$

$$V = \frac{(D_c)^2\ RT_{60}}{(.03121)^2\ Q}$$

Fig. 2

First obtain an accurate measurement of the RT_{60} of the room. This is best done by both sending the signal in octave bands and receiving in octave bands. Let's suppose that for an example we measure an RT_{60} at 2000Hz of 2.5 secs. Further let's assume that our test loudspeaker has a Q = 5 at 2000Hz. We next proceed to measure D_C at 2000Hz.

At 8 feet on axis we measure 100 dB-SPL and at 100 feet we measure 88.2

$$D_C = 8' \times 10^{(100-88.2/20)} = 31.12'$$

and

$$V = \frac{(D_C)^2 RT_{60}}{(.03121)^2 Q} = \frac{(31.12)^2(2.5)}{(.03121)^2 5} = 497{,}120.5\ ft^3$$

The actual space was 500,000 ft^3, thus the error in calculation is 0.6%. Anytime any acoustical data falls within 1% you have a precision measurement.

To anyone who has ever been faced with an intricately cut up, Gargoyle decorated, hidden recesses cathedral, for example, this method is heaven sent. In small "dead" rooms, the accuracy of this method will, of course, suffer as it is based upon the classic Sabine equation. However, small "dead" rooms are normally easy to measure in the conventional way.

SANDLAPPER SOUND CHURCH BROCHURE

We see many outstanding sound contractor brochures aimed at the churches and have often wished we were able to reproduce a brochure in our Newsletter. DARYL NATIONS and BOB MAGEE of Columbia, South Carolina were in our Orlando class in 1977. There are a few problems in reproducing a brochure and the message may not come across in this reproduction but we wanted to try to reproduce the front and back cover of what we think is a beautiful little brochure.

TOP

BACK

SANDLAPPER SOUND:

- **Years of experience in radio engineering, broadcasting and sound reinforcement**
- **Featuring world famous ALTEC equipment**
- **We do our own installations including cabinetry to sound equalization**
- **With a special outreach to Latin America Missions**
- **Visit some of our recent installations in the Columbia Area**

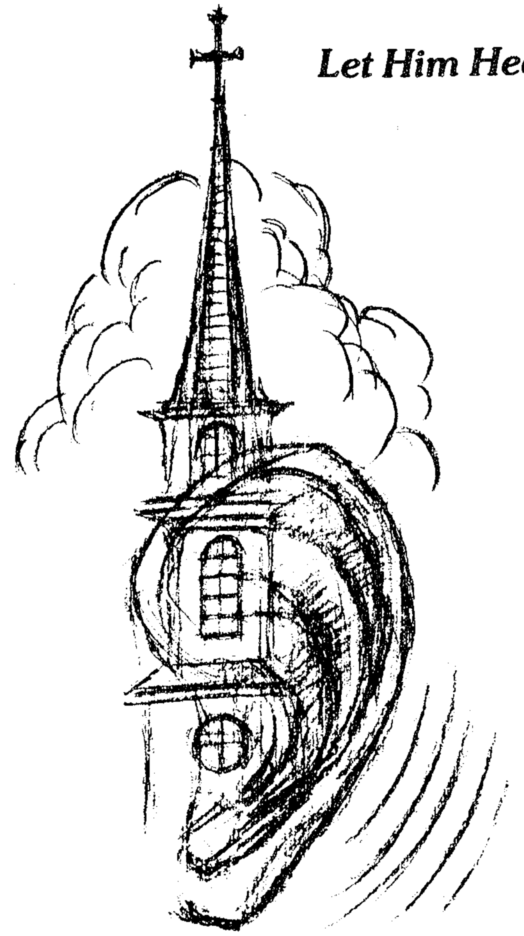
Columbia Bible College
Multi Audio System in the Student Center
Columbia's First Baptist Church
Dunn's Chapel
Taylor Evangelical Methodist
Ridgecrest Baptist

CENTER FOLD

FRONT

He who has an ear to hear . . . .

Let Him Hear



But can he hear in your church?

BOTTOM OF BROCHURE

THE "FLOW OF CURRENT FROM POSITIVE TO NEGATIVE"

Referring to the Klipsch Letter to the Editor of the IEEE reproduced on Page 10 of the last Newsletter (Vol 4 # 4) we found in a book (new to us) what we believe is the answer. This book (reviewed in this Newsletter in Book Reviews) *Introductory Circuit Analysis*, second edition, states on page 76,

> By *convention* the direction of I as shown in Fig 5.1 is *opposite (italics mine)* to that of *election flow*. The reason for defining conventional flow in the opposite direction stems from an assumption made at the time electricity was discovered, that the positive charge was the moving particle... By following the direction of conventional flow we notice that there is a rise in potential across the battery (- to +), and a drop in potential across the resistor (+ to -)"

Observing these effects in the early measurements of electricity could easily have led to the false conclusion.

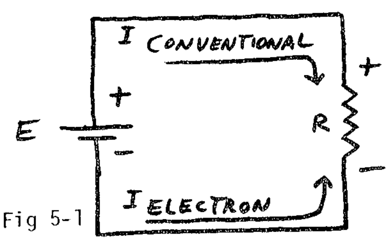


Fig 5-1

SYNERGETIC AUDIO CONCEPTS

LOGS AND DB

by
Cecil Cable

For those with inquisitive minds, consider this:

$$10^{2.5} = 10^{\frac{5}{2}} = 10^{5\times\frac{1}{2}} = (10^5)^{1/2} = \sqrt[\frac{1}{1/2}]{10^5} = \sqrt[1\times\frac{2}{1}]{10^5} = \sqrt[2]{10^5} =$$

$$= \sqrt[2]{100{,}000} = 316.227$$

$$10^{2.5} = 10^{5/2} = 10^{5\times\frac{1}{2}} = (10^{1/2})^5 = \left(\sqrt[\frac{1}{1/2}]{10}\right)^5 = \left(\sqrt[1\times\frac{2}{1}]{10}\right)^5 = \left(\sqrt[2]{10}\right)^5 =$$

$$\sqrt{10}\times\sqrt{10}\times\sqrt{10}\times\sqrt{10}\times\sqrt{10} = 316.227$$

$$10^{2.5} = 10^{2+0.5} = 10^2\times 10^{0.5} = 10^2\times\sqrt[\frac{1}{0.5}]{10} = 10^2\times\sqrt[2]{10} = 100\times 3.16227$$

$$= 316.227$$

The rule:

The reciprocal of an exponent of a base may be used as the root of the base without a change in value (of the antilog)

$$10^{.5} = \sqrt[2]{10}$$

The reciprocal of a root of a base may be used as the exponent of the base without a change in value (of the antilog)

$$\sqrt[2]{10} = 10^{.5}$$

Exponents of a base, multiplied together are equivalent to the antilog of one exponent raised to the second exponent

$$10^{2\times3} = 100^3 \text{ or } 1000^2$$

One exponent of a base, divided by a second exponent is equivalent to the antilog of the first exponent taken to the root of the second exponent.

$$10^{4/2} = \sqrt[2]{10{,}000}$$

One exponent of a base, divided by a second exponent, is the equivalent to the second exponent taken as the root of the base, raised to the first exponent.

$$10^{4/2} = (\sqrt[2]{10})^4$$

THE j OPERATOR AND IMPEDANCE NOTATION

RECTANGULAR	POLAR	TRIGNOMETRIC	EXPONENTIAL
$a \pm ib$	$r\angle\theta$	$r(\cos\theta + i\sin\theta)$	$re^{i\theta}$
$R \pm jX$	$Z\angle\theta$	$Z(\cos\theta + j\sin\theta)$	$Ze^{j\theta}$
$2.5 + j4.33$	$5\angle 60^\circ$	$5(\cos 60^\circ + j\sin 60^\circ)$	$5e^{j1.05}$

$$e^{ix} = (\cos x + i\sin x)^*$$

* X IN RADIANS - TRIG. FUNC. PERFORMED IN RADIAN MODE

NOTES

$$R + jX = \sqrt{R^2 + X^2} = Z \qquad \tan^{-1}\frac{X}{R} = \theta$$

$$Z\cos\theta = R$$

$$Z\sin\theta = X$$

DEGREES OR RADIANS MAY BE USED

$$\frac{R}{\cos\theta} = Z$$

$$\frac{X}{\sin\theta} = Z$$

DEGREES OR RADIANS MAY BE USED

HOSPITAL SAFETY

From ED LETHERT, Independent Consultant, Minneapolis, Minn: I just picked up the latest Woodhead catalog. They have an interesting new test unit for hospitals. It detects both leakage currents and static charge. (If you haven't written for a copy of the Woodhead catalog - written up in the last Newsletter - you should. Address: Daniel Woodhead Company, 3411 Woodhead Drive, Northbrook, ILL 60062.)

H.P. has an interesting application note (AN 718) titled, "Patient Safety". It has some very interesting information on current flow through the human body and circuit faults that can cause these currents to flow.

COMPLEX IMPEDANCE CURVES

Many have asked about the complex impedance curves Dick Heyser shows in his tests of loudspeakers in AUDIO Magazine. The illustrations reproduced here illustrate how impedance magnitude (Z) vs frequency can, by measuring phase angle (Θ), convert each variation on the Z plot into its resistive and reactive components.

Therefore, on Fig. # 1 the steepest part of the curve marked (1) is reactive and is essentially acting like an inductance (Z increases with increasing frequency). At (2) the Z is acting like a capacitance (Z decreases with increasing frequency). At (3) and (4) the same type of changes are occurring.

On Figure # 2, Point (1) is plotted as an ac resistance (ACR) and an inductive reactance (X_L). Reading the graph as closely as we can we find that for Point (1) on Figure # 2, ACR = 19Ω and X_L = 13.5Ω, therefore,

$$Z = \sqrt{(19)^2 + (13.5)^2} = 23.31\Omega$$

$$(\Theta) = \mathrm{Tan}^{-1}\frac{13.5}{19} = 35.39^{\circ}$$

Point (2) on Figure # 2 is of more concern as it looks capacitive and the Heyser report stated that the Z = 12Ω at 1800 Hz with $\Theta = 42^{\circ}$ (a voltage lag)

$$12\angle 42^{\circ} = 8.92 - j8.03$$

Which means that the X_C = 8.03Ω.

Points (5) and (6) reveal other resistance points (curve is flat at these points) and the difference between the ACR and these values is primarily the motional impedance Z_M (a loudspeaker generates voltage as it moves as well as uses voltage)

Thus it can be seen that the "Complex Impedance is simply a different way (rectangular coordinates) of plotting magnitude and phase angle (polar coordinates.)

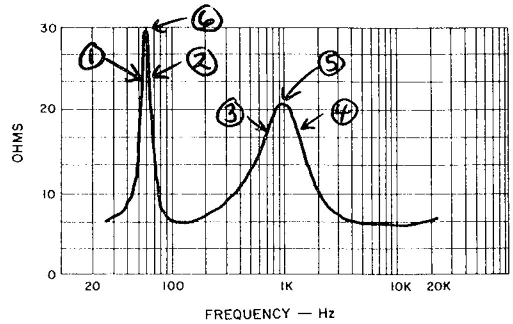


Fig. 1—Magnitude of impedance.

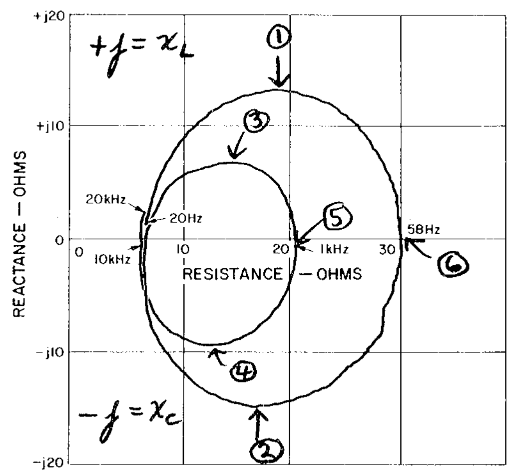


Fig. 2—Complex impedance.

HELMHOLTZ RESONATOR CALCULATIONS

The resonant frequency of a Helmholtz resonator can be found by:

$$f_r = \frac{C}{2\pi}\sqrt{\frac{S}{\ell V}}$$

Where f_r is the resonant frequency in Hz

C is the velocity of sound in M/sec (typically 343 M/sec) or in ft/sec (typically 1130 ft/sec)

S is the cross sectional area of the resonator's neck in M^2 or ft^2

ℓ is the length of the resonator's neck in M or in ft.

V is the cavity volume in M^3 or ft^3

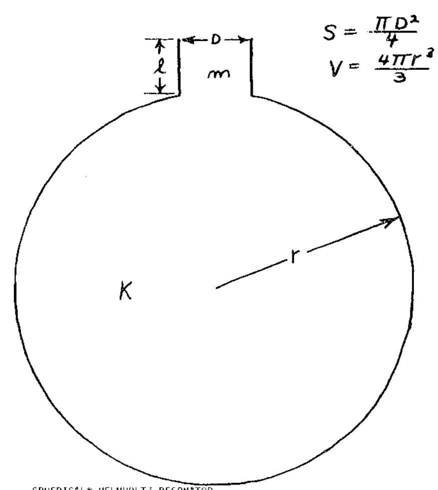
A Helmholtz resonator acts as an absorber of sound by coupling a mass (m) in the neck of the resonator to the air cavity which acts as a spring (K). Energy is dissipated at resonance by the motion of the air in the neck of the resonator.

A most useful variation of this equation is the ratio of the neck cross sectional area to the neck's length:

$$\frac{S}{\ell} = \frac{(f_r)^2 V}{\left(\frac{C}{2\pi}\right)^2}$$

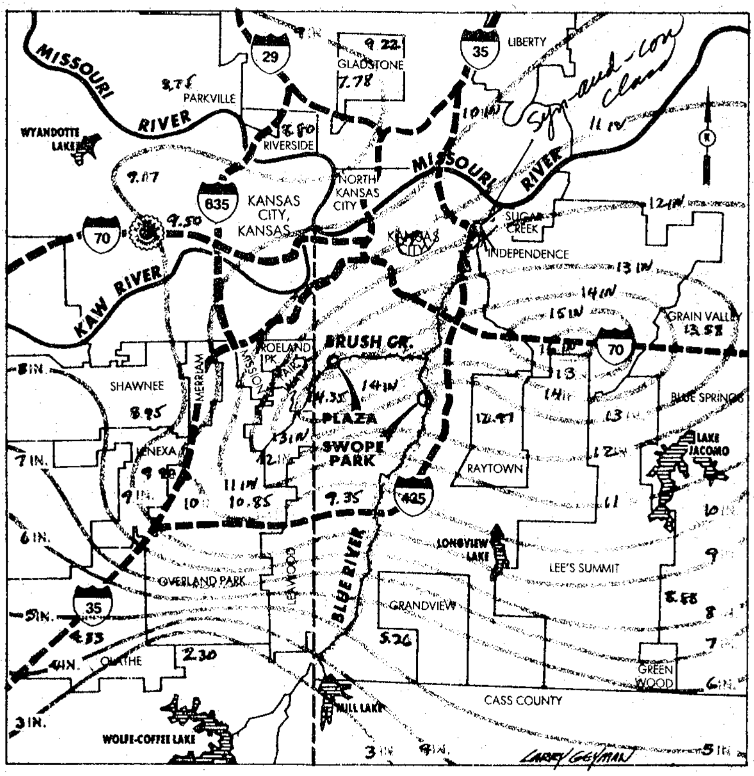
If, for example, you wish to "tune" a concrete block by putting a slit in it, the cavity volume and neck length are already predetermined by the manufacturer of the block. This leaves the cross sectional area of the neck as the logical parameter to adjust.

$$S = \frac{(f_r)^2 V\ell}{\left(\frac{C}{2\pi}\right)^2}$$



SPHERICAL* HELMHOLTZ RESONATOR

*May also be in cubical shape



We asked on our Suggestion Sheet if our Kansas City hotel was convenient by car (class held September 14-16). One member of the class said, Yes, but more convenient by boat.

How Much Rain? **The center of storm activity Sept. 12-13 was in eastern Jackson County where 16 inches of rain fell. The storm moved outward toward the Plaza where 14 inches fell to its weakest point in the south where 3 inches were recorded.**

MOS op amps form 'pink noise' source

by John Maxwell
National Semiconductor Corp. Santa Clara Calif.

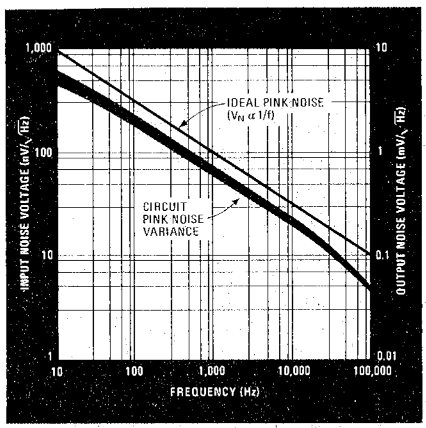
Many amplifier and acoustical measurements need a "pink noise" source—that is, one having constant noise power per octave, in contrast to a white-noise source, which has constant noise power per unit bandwidth. Such a generator can be built using a pair of MOSFET-input operational amplifiers, as well as a few passive components.

The circuit uses the low-frequency noise characteristics of the metal-oxide-semiconductor field-effect transistor (pink noise is not usually produced by complementary-MOS devices). The transistor eliminates critical components and the "pinking" filter typically required in lab units. It is significantly lower in cost than commercial noise sources.

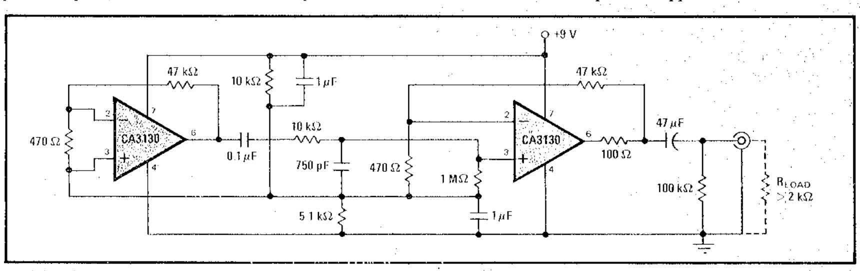
The circuit, shown in Fig. 1, uses two cascaded CA3130 amplifiers with a composite gain of 10,200. Generally, any MOSFET amplifier can be used to produce similar characteristics to those plotted in Fig. 2.

The spectral output of the noise circuit closely follows the ideal pink-noise curve. The output noise is 16 millivolts per octave. The total output is 70 mV rms (420 mV peak to peak), which should be adequate for most circuits needing this noise generator.

Total current drain is less than 5 milliamperes when the circuit is operated from a 9-volt battery. This allows nearly 100 hours of intermittent operation, making the noise source ideal for portable applications. □



2. Noise-source output spectrum. For adjustable noise output of 7 to 140 mV, replace the 47-kΩ resistor in second stage with a 100-kΩ potentiometer in series with a 4.7-kΩ fixed resistor.



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1. Pink-noise generator. An RC low-pass filter is inserted between the two amplifiers to compensate the upper audio band noise and to eliminate unwanted high-frequency noise. Each amplifier's stage provides a gain of about 101.

Electronics/March 31, 1977

BOOK REVIEW

INTRODUCTORY CIRCUIT ANALYSIS. During our travels through New England each year, one delightful interlude is spent with the Glen Ballous in Connecticut. Glen and I devote at least one day to searching old book stores in the area for used technical books. This year we uncovered what we feel is the *best book* on the subject we have seen.

Introductory Circuit Analysis, 2nd edition by Robert L. Boylestad and published by the Charles E. Merrill Co., Columbus Ohio 1972, is lavishly illustrated with correctly worked out examples. The chapter on "Sinusoidal Alternating Current" is replete with valuable information ranging from a complete step-by-step derivation of how to handle RMS calculations of complex waveforms to (ELI the ICE man - E leads I when L is present and I leads E when C is present.) 827 pages with 26 chapters that really answer basic questions with real meat.

I don't know if this book is still available but if it is it belongs in every audio library. Try to find it in new or used book stores.

THE DEFINITIVE GUIDE TO THE ELECTRONICS OF MUSICMAKING by Alan Douglas (TAB Book # 832) $9.95 Hardbook $6.95 paperback.

This huge guidebook is a thorough work on the theory, design, and practical technology of electronic musical instruments with special emphasis on the organ. It covers the whole spectrum of electronic music synthesis: the latest advances and the early time-proven techniques.

There is complete info on conventional multinote instruments, tone forming, loudspeakers, power supply units, vibrato circuits, and keying. There is complete data on how each part works--in theory and in actual commerical instruments, to help one gain a nuts-and-bolts understanding of the workings of every organ made.

Many commerical electronic instruments are thoroughly treated, including all their pertinent schematics and specifications. There's a whole Chapter on experimental methods, showing how to expand and adapt reliable circuits and devices so the reader can build his own customized organ right from the ground up.

We found it interesting and useful.

***

IMPEDANCE by Rufus P. Turner, published by Tab Books No. 829 at $5.95, softback 195 pages.

An excellent basic book for self-teaching. This book brings the reader through the basic definitions, applications and measurements of impedance. It covers all primary cases of Z using rectangular and polar notation with just a touch of the trignometric forms. Easy to read.

OF INTEREST

ROLLING STONE recently published their annual High Fidelity issue in which they list the equipment that they feel is best within a given price range. The large speaker systems are enjoying a renaissance. The Klipschorn, the Bozak B-310 and the JBL L212 are listed as best equipment in the two most eclectic systems. Anyone around 25+ years ago when <u>The Wall Street Journal</u> called the Klipschorn the King? Our high futility shop in Lafayette, Indiana featured the Bozak B-310 and the Klipschorn. It's good to see the greats still with us.

*****

POPULAR SCIENCE, August 1977: Bubble Memory. The first commercial product to use those strange magnetic bubbles we told you about in the February 1970 issue - when they were only laboratory curiosities - has been introduced by Texas Instruments. It is a 17-pound portable computer terminal that can store the equivalent of 20 pages of single-spaced typewritten text in its bubble memory. Not only do the magnetic bubbles store a fantastic amount of information on a tiny chip, they do not forget when the power is turned off. An operator can carry the terminal around - switching it on to record data as needed - then at the end of the day dial a central computer and transmit the data over an ordinary telephone line. (Courtesy Nick Metal, Vancouver, B.C.)

*****

POPULAR ELECTRONICS: *SPEAKER SYSTEM MEASUREMENTS -- IS PHASE RESPONSE IMPORTANT?* by Julian Hirsh. "The reverberant response of a speaker system is close to being the analog of its perceived (subjective) frequency response." ???????

*****

ROLLING STONE: (Interview with Sidney Harman) Sidney Harman did not set out to become a prominent "humanist executive." He was born in New York in 1918 and after graduating from City College of New York, landed a job as an engineer with the David Bogen Company, *a fledgling sound company that no longer exists. (Italics mine.)*

*****

THE CHRISTIAN SCIENCE MONITOR: *THE PLANET'S FRIENDLY GENIUS.* Bucky Fuller says, "You don't have to know anything to be negative. To be positive you really have to know something."

*****

NOISE/NEWS: Just six years ago, in December 1970, the U.S. Environmental Protection Agency came into being. Now the nation's largest regulatory agency, EPA, employs more than 9,000 people and its overall budget exceeds $2 million each day.

*****

LOS ANGELES TIMES: *ELECTRONICS INVADES MUSIC CLASSES.* Musicians are by nature a rather conservative bunch, says Robert Klotman, the chairman of the Music Education Department at the University of Indiana...Our studies have shown that it *takes 50 years for a new idea to permeate the American school system.* But in the last seven or eight years we have had to accommodate dramatic changes.

...continued next page

OF INTEREST, CONTINUED

ELECTRONICS: Radio Shack will throw the retailing muscle of its 6,000 stores into the home-computer market. It plans to introduce a machine of its own design and construction reportedly built around a Zilog Z-80 microprocessor chip. It will sell for about $800 and includes a cathode-ray tube, 4,000 bytes of memory, a typewriter-style keyboard and a cassette interface.

*****

THE CHRISTIAN SCIENCE MONITOR: *CALCULATORS FOR KIDS "ADD UP"*. The calculator challenge is also forcing educators to face the fact that they really don't know how best to teach elementary mathematics and that incisive research into this subject is urgently needed. Dr. Bell (University of Chicago educator) says that he did not find undue reliance on calculators when the machines were tried in classes with 20 different teachers. He feels this concern is being used as an excuse to avoid facing up to the real calculator challenge - that of teaching the ability to use math and not just to add, subtract, multiply and divide.

*****

MONTEREY PENINSULA HERALD: *CALCULATORS IN SCHOOL.* When a first or second grader can quickly find a square or cube root, it puts a different light on the learning sequence. By pushing a button, children can solve decimal problems as easily as they deal with whole numbers. At an early age, they can be taught to solve interesting problems, without having much skill at basic arithmetic.

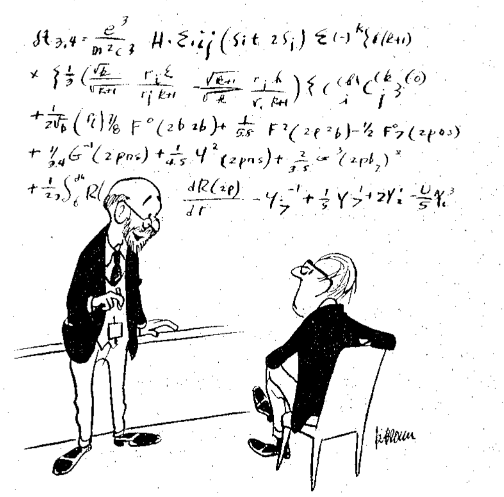
*****

NOISE CONTROL ENGINEERING: *NEW TRANSPARENT NOISE-BARRIER REDUCES NOISE, PERMITS VISUAL MONITORING.* Immediate availability of a new limp, flexible, transparent film that resists the passage of sound waves and dramatically reduces noise transmission is announced by Ferro Corporation's Composites Division. It is marketed under the trade name of Coustiview. It reduces noise from 15 dB to 21 dB (STC), yet permits visual monitoring of machinery and equipment, and it is recommended for use as hanging curtains and as "windows" in opaque curtains and fabricated enclosures for noise control systems where visual monitoring is desired. For more information, write Ferro Corporation, Composites Division, Noise Control Products Department, 34 Smith St., Norwalk, Connecticut 06852.

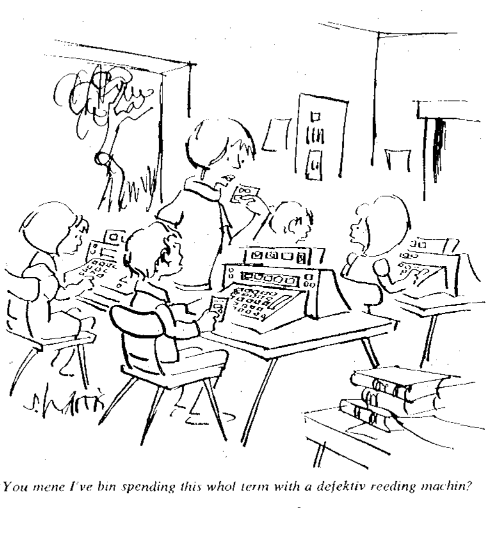
GRIN

The cartoon brings to mind a couple of stories that we heard recently and have been sharing in our classes. Every Fall we are near Greenville, PA and always stop over to see our friend and many-time graduate, FLOYD COOPER and often we share dinner with Dr. Brown, head of the Physics Department at Allegheny College who took classes from Dr. Norbert Wiener. Dr. Brown tells of Dr. Wiener filling a black board with equations and as he was starting on the second board, he said, "it is obvious...." A young student stopped him with, "Dr. Wiener, is it obvious?" Dr. Brown said that Dr. Wiener stared at the board of equations for a full 10 minutes, turned back to the young man and said, Yes, it is obvious."

When RALPH GIBSON of Gibson Associates in New Hartford, Conn. (another many-time graduate) heard the story, he told us about one of his classes at MIT. A young doctorate aged 19 taught his sophomore physics class. The professor wrote on the blackboard with his right hand and erased with his left hand as he moved across the board.

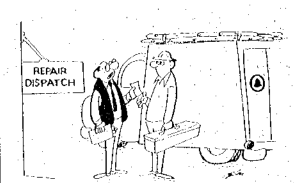


"It's just a guess, of course, but at least it's an educated guess."



You mene I've bin spending this whol term with a defektiv reeding machin?

Sent in by
Tom McCarthy



"The lady who called in the trouble report says that her bathtub fills up every time the telephone rings."

ARTICLES OF INTEREST

EFFECTIVE ACOUSTIC CENTER REDEFINED is a Letter to the Editor of the J-ASA by W. James Trott. It is the most erudite discussion to date of how to calculate or measure the acoustic center of a source at varying frequencies.

The K numbers used in this article are defined as

$K = 2\pi f/c = w/c = 2\pi/\lambda$

Where f is the frequency in Hz
c is the velocity of sound in ft/sec
λ is the wavelength in ft.

(Trott uses but does not define this term (K) in his letter.)

This paper is a basic one of value to transducer engineers. System engineers avoid this problem by using a reference distance far enough from the source to "swamp out" errors caused by mislocating the acoustic center. Measuring at 10' you could miss the acoustic center of a 12" cone by 1 radius and only have an error of

$$20 \log \frac{126''}{120''} = .42 \text{ dB}$$

This 10' data is then normally reduced to 4' for publication.

For those wishing to study the magnitude and detail of the acoustic center problem, we highly recommend this article. It appears on pages 468-469 of the J-ASA Vol 62, No 2 August 1977.

*****

REVIEW OF NOISE PROPAGATION IN THE ATMOSPHERE. This 16 page tutorial paper in the J-ASA written by J.E. Piercy, R.F.W. Embleton, and L.C. Sutherland presents a general review of sound propagation outdoors. Some of the areas covered are geometrical spreading, atmospheric absorption, ground effects, refraction, the effect of atmospheric turbulence, and topographical effects. It includes a bibliography with 96 references from the literature on the subject. Many areas needing only careful experimentation in order to confirm or deny theoretical equations currently available but as yet unverified are identified.

This interesting and useful article appeared in the J-ASA Vol 61, No 6, June 1977 on pages 1403-1418.

*****

TIME ALIGNMENT™ IN LOUDSPEAKERS by Edward M. Long. First of all, let me state that I am a rabid enthusiast for Time Align™ as done by Ed Long on the UREI monitors. Secondly, I also believe that each listener must prove the effect for himself. Finally, I highly recommend reading Ed Long's article in the August 1977 issue of AUDIO Magazine. The article contains the basics necessary to begin *thinking* about the concepts. Then, all that remains is to hear the concept *properly* demonstrated.

*****

CLASSIFIED

NOT A GRADUATE BUT A SURVIVOR

DAVE OGDEN of Com-Tec in Appleton, WI (Kansas City 1977) remarked when we passed out certificates at the end of the class, "I'm not going to call myself a graduate but a survivor." I hope the Kansas City flood had something to do with that statement!

******

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