"SYN-AUD-CON"
SYNERGETIC AUDIO CONCEPTS

P.O. BOX 1134, TUSTIN, CALIFORNIA 92680

newsletter

VOLUME 5, NUMBER 3
APRIL, 1978

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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VOLUME 5, NUMBER 3

IVIE ELECTRONICS NEW SYN-AUD-CON SPONSOR

Ivie Electronics, well known in our industry for their pioneering efforts in handheld real time analyzers, has become Syn-Aud-Con's most recent sponsor.

Located in the mountain West at Orem, Utah, they have just recently built a new 30,000 ft^2 plant on a 9½ acre site near the 11,750 ft. Mt. Timpanogos.

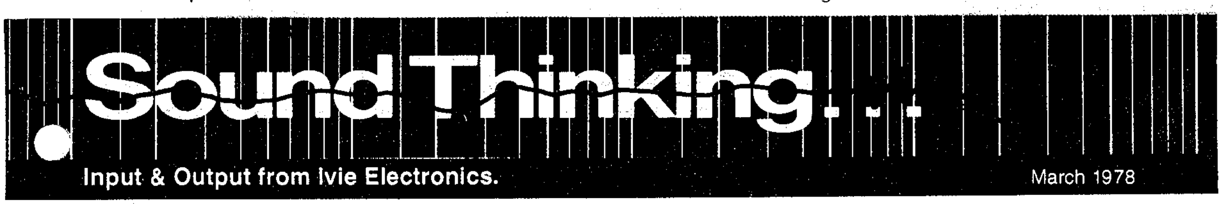
One of the major attractions Ivie Electronics had for Syn-Aud-Con was the consistently high caliber of personnel they are bringing together, the most recent being Bill Raventos (formerly of EV).

A second distinguishing feature has been the remarkable desire on their part to communicate about any product problems they have experienced in their production and their instant correction of such problems when the user returns the unit to them for inspection. At first we thought they were responding with such interest because of our contact with so many Syn-Aud-Con graduates, but graduates owning Ivie products have enthusiastically endorsed Ivie for exactly the same kind of performance on their behalf.

Ray Ivie's company is one that we are looking forward to working with closely. He's done an unusual job of attracting extremely talented people to a beautiful part of the United States to work on advanced innovative projects. It all sounds synergetic.

"SOUND THINKING" BY IVIE

Ivie Electronics has published their first Newsletter called Sound Thinking.



By writing the address shown below you can receive Ivie Electronics' new Newsletter free of charge.

Bill Raventos tells that they printed a few issues with the "Merry notes and dozy quotes from the Little Lambs at Ivie" on the front of the Newsletter and put them on Ray Ivie's desk. Bill says that Ray Ivie said nothing, being a very quiet man of few words. He just went home for the rest of the day.

Merry notes and dozy quotes from the Little Lambs at Ivie.

We'd be willing to predict that much of interest should appear in this publication as their experience grows in the instrumentation field. Few market places are as volatile and competitive today as audio instrumentation and the competition extends well beyond the "hardware" being offered by manufacturers into the "software" of clever, efficient, and accurate application techniques.

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ANAHEIM - SEATTLE - SAN FRANCISCO

SYNERGETIC AUDIO CONCEPTS

1978 SEMINAR SCHEDULE

If we can get our act together by Fall, we will start 4-day seminars with the 1st day for new attendees, with graduates coming in for three days. The first day will be "basics" and will be conducted by knowledgeable engineers from our sponsors who are especially good at communicating technical information.

Our full schedule is still tentative but most likely we will follow this schedule:

Anaheim, CA - January 31-February 2
Seattle, WA - February 14-16
San Francisco, CA - February 22-24
Vancouver, B.C. - April 18-20
Los Angeles, CA - May 10-12
Chicago, IL - September 12-15
New York, NY - September 26-29
D.C. - October 3-6
Atlanta, GA - October 17-20
Orlando, FL - November 14-17

Rauland-Borg Company is sponsoring three Syn-Aud-Con seminars in St. Louis, Orlando and Los Angeles during the Fall for their dealers.

REVIEW OF DICK HEYSER'S ARTICLE IN AUDIO MAGAZINE--ALTERNATIVES

I find Dick Heyser one of the fundamentally inquisitive minds in audio today. His recent article in Audio Magazine entitled, *Alternatives*, certainly serves its intended purpose of generating new thoughts on old subjects.

One of the most provocative statements in the article is:

> What is melody, or even a melodic contour? Stretch the mind a bit. If each allowable tone is assigned as a dimension, then certain groups of tones, *bearing particular relations to each other (italics mine)*, define subspaces of finite dimensionality. These subspaces may be combinable in a different manner so as to form characteristic patterns which have extremum metric properties relative to subspaces formed from random combinations of tones. That is, the preferred subspaces are more densely packed with less distance separating members of the subspace.
>
> ....The conceptual "distance" between certain notes, and I do not mean where they are on the musical scale but whether they seem to "fit" together, seems to form an attractive way of discussing chords and how they might fit together in the various combinations we might *think* of as melodies.

Dick further makes a remark, earlier in the same article, that

> I contend that the thing we call distortion in audio, both objective and subjective, can be regarded as a warping of the geometry within a given frame of reference.

Dick also alludes to the observer-observed effect encountered in conventional instrumentation particularly with regard to the trade-offs between filter resolutions and filter transient response.

What, I believe, is being compared here is Mother Nature's, to all intents and purposes, perfect analyzer--a gifted human being with above normal hearing and mental processes and the electro-acoustic tools developed to date. Where better to search than in the domain of this "perfect analyzer" though this may be cybernetic.

Some of the "warpings" are obvious such as the inability of any loudspeaker to reproduce the polar response of any conventional musical instrument. The electronic guitar can be imitated but not the acoustic guitar. Reduced to conventional terms, the polar response varies from the original in excess of ±10 dB in most cases.

Some less obvious abilities of this perfect analyzer have been described recently in the JASA. Tadanobu Tsunoda, the developer of the Cerebral Dominance Key Tapping Test as a tool for the evaluation of pathological cerebral conditions, has found that Japanese hear both Japanese speech and Japanese music quite differently than do Westerners. (See write up in this Newsletter of *Do you Hear the Singing of the Cicadas*, Letter to the Editor, JASA by Birtin & Kearsley.) Japanese hear Western music in much the same way as Westerners. For example, the left cerebral hemisphere is normally verbal and the right hemisphere responds to music, noise, etc. For Westerners, consonants in speech excite the left hemisphere predominately, while vowel sounds--music, noise, etc.,--predominate in the right hemisphere. For Japanese, vowel sounds predominate in the left hemisphere, consonants in the right, Japanese music and insect noises in the left but Western music in the right.

Tsunoda asserts *that these differences between the two groups are attributable to a difference in the perception mechanism of complex sounds*. He concludes that there is a special switching mechanism operating in vowels which for the Japanese often have meanings beyond the simple letter associations. That is, vowels themselves are language connotative and frequently a single vowel will have complete and diverse meanings. On the other hand, for the Westerner the vowel is a mere function of a thought pattern, the basic unit of language being syllables.

Here is both a testing method, for the "perfect" analyzer, and a set of highly suggestive questions.

In the majority of Western listeners, for example, is distortion or melody in the same or different cerebral hemispheres? Tsunoda implies they are treated the same in Westerners, and in critical listeners, could distortion go to the verbal hemisphere (the left) or for trained musicians perhaps there is a shift in cerebral dominance?

Might it not be possible to explore if our brain is assigning differing cerebral dominance to speech and music; and noise might not be doing so because the different hemisphere provide differing analysis of the signal and that some higher order brain function then samples the various analysis with the resulting fascinating differences in subjective perception so often noticed and commented on in human beings.

Let's have more of Dick Heyser's "Audio Wan Kanobi" approach to the audio "force". Ideas must come before material manifestations are possible. All the materials necessary to construct a jet airplane were present on this globe at the time of the caveman - just the ideas necessary to use them were missing. Dick's peeks out of the cave are refreshing.

DBD

REVIEW OF "DO YOU HEAR THE SINGING OF THE CICADAS--ASA JOURNAL

We have frequently mentioned in class, during the discussion of the Peutz articulation loss equations, that Japanese speech is vowel centered and Occidental speech is consonant dependent.

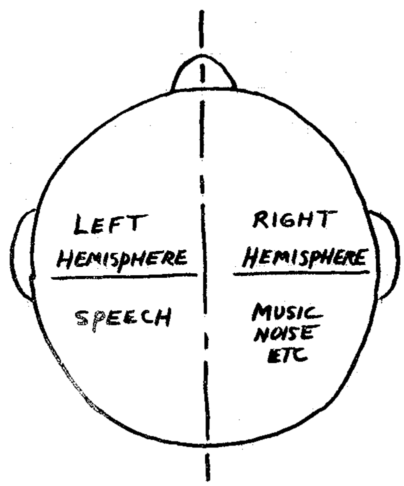
In a recent edition of the Journal of the Acoustical Society of America, vol 62, no. 4, October 1977, pages 1061-1062, A Letter to the Editor from M. A. Birtin and E. A. Kearsley, entitled *Do You Hear the Singing of the Cicadas?* discusses the work of a Japanese research worker, names Tadanobu Tsunoda, who developed the Cerebral Dominance Test.

The Cerebral Dominance Test is a test wherein the subject being measured can be examined to determine which cerebral hemisphere of the human brain has dominance while listening to different sounds:

> As with many bodily functions, the left ear corresponds to the right cerebral hemisphere, etc.
>
> Using as a sound (1) a recording of a steady pure (A) vowel tone, (2) a 1 KHz pure tone, and (3) white noise....Tsunoda found that for 72% of normal Japanese subjects the test showed the typical pattern of left hemisphere dominance for the vowel sound (which is usual for speech) and right hemisphere dominance for the pure tone and the white noise (which is usual for machine noise and orchestral music).
>
> He....tried his new method on a number of Westerners resident in Japan with surprising results - a distinct dominance effect of the steady-state vowel (A) in the nonverbal hemisphere was found for Western subjects.

TOP VIEW OF HUMAN HEAD

Left ear connected to right hemisphere



Right ear connected to left hemisphere

Typical hemispheric division of auditory perception of Westerner

> What the results seem to suggest is that Westerners perceive a steady-state pure vowel sound as a noise (or perhaps music), while to Japanese it is a verbal sound. Tests with foreign born Japanese (who had been brought up with the local language) were comparable to those of Westerners as were those of Chinese and Korean subjects, so that the effect is apparently neither genetic nor Oriental in essence.
>
> The puzzling difference also emerged in dominance for emotive and animal sounds, the native Japanese "hearing" them in the left or verbal hemisphere whereas the others "heard" them in the right half of the brain. Even stranger, the Japanese "hear" traditional Japanese instruments on the verbal side and Western music on the other. Westerners "hear" all music on the same side. (ED: Westerners normally hear consonant sounds in the verbal or left hemisphere.)

How many questions this report raises. Do those musicians who "play by ear" use a different cerebral dominance order? Do drugs effect cerebral dominance order? Does very high level change dominance order? Is opera music in a foreign language (not understood by the listener) right or left cerebral dominance? Is a racing motor in the verbal hemisphere for a master racing mechanic? Is skiing, horseback riding, shooting, etc., controlled by one hemisphere in an amateur and another in the case of exceptionally gifted experts?

Since it is demonstrative that what we perceive as reality is dependent upon the human mind's processing, what programming of the young infant causes what adult orientation and even more intriguing, which programming might conceivably be desirable or would uniformity of perception be desirable?

How fortunate we are to live amidst such interesting, unsolved, and yet investigable questions.

DBD

TI SHIPS 140-K BUBBLE CHIPS

Mind boggling - "TI ships 140-K bubble chips to Air Force".

The new 400 mil. square chip uses a block replicate structure instead of a major-minor loop configuration. Intended for use in replacing airborne disk and drum memory systems, it is speculated that TI may turn to it for future commercial products as well.

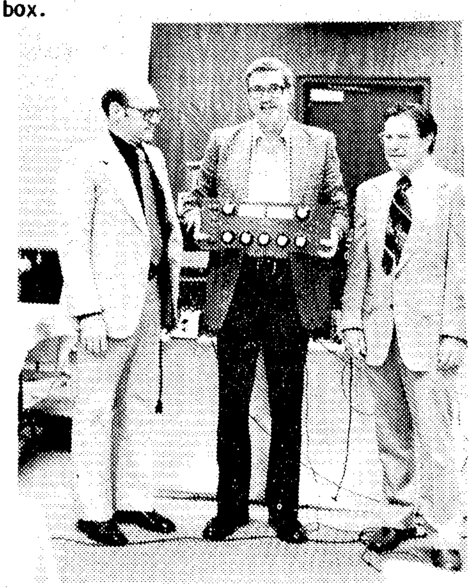
With densities getting this high, I suspect that de-bugging such systems will require a couch.

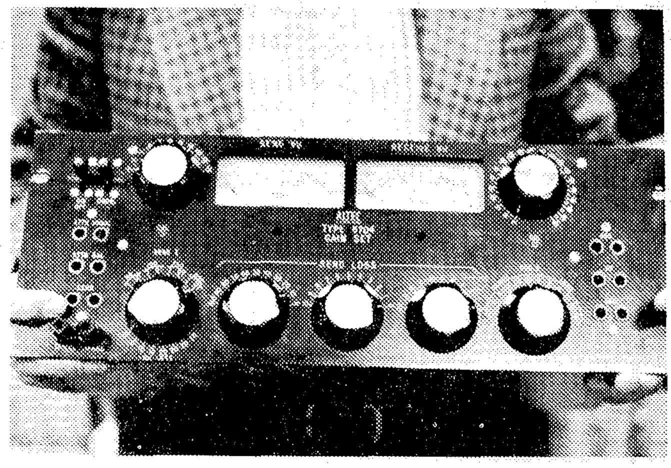
DBD

A PLEASANT SURPRISE--A GIFT FROM ALTEC

One of the most pleasant experiences in life is friendly recognition by ones peers.

At the Anaheim, CA class (February 1978), I was busy talking to the class just before lunch the third day of class when two men entered the classroom carrying a large package bound with ribbons and marched up the aisle to hand me the box.





Photos by Fred Fredericks, San Diego

The two men, Bob Davis (L) and Paul Spranger (R), both executives at Altec, had located and rehabilitated an Altec 9704 gain set designed by Art Davis when I first worked for him. They presented the unit to me as a gift expressing their's and Altec's regard for my past association with the company many years ago. The picture on the right is a close-up of this classic measuring tool. If you have ever wondered what I look like when totally surprised and pleased, the picture above conveys it fully.

DBD

MORE ON AZIMUTH ALIGNMENT OF TAPE RECORDERS

Further details on using the real time analyzer for the adjustment of tape recorders came in from Bill Raventos of Ivie.

> I've found that the best type of test tape to use in conjunction with a real time analyzer for adjusting playback equalization is a standard sweep tape provided by many of the instrumentation tape manufacturers (ours was ordered from STL). Basically, the tape contains a swept sinewave, sweeping upwards from 500 Hz to 20 KHz, with one sweep occurringeach one hundred milliseconds. By putting the IE-10A into the slow integration mode (going from C-weighting to octave), or by using the IE-30A on D3 it is possible to see an almost continuous display on the screen from which playback EQ adjustments can easily be made. After playback EQ has been set up using this swept sinewave tape I normally then inject pink noise from our pink noise generator into the input of the tape machine (taking care to set up the record level on blank tape so as not to saturate the tape with the high crest factor of pink noise) and adjust the record EQ with the pink noise so that it is a direct copy of whatever the final playback EQ was. This insures flatness from recorded playback and optimal performance throughout the machine.
>
> With regard to the azimuth adjustment, it appears that I neglected to consider tape machines with unbalanced outputs! (Newsletter Vol 5, No 2, page 18) I never have used unbalanced machines, but obviously electrically reversing polarity from one channel to the other shorts both active outputs to ground and if you can get any reading on your real time analyzer it will be hum and noise only. So my little addition for adjusting azimuth will work only with a machine with balanced output, or by taking the signal directly from the playback heads.

It surely is apparent to us that lack of a real time analyzer today is as like not owning a VOM forty years ago.

CALIFORNIA DROUGHT IS OVER?

The California drought, that two year period that proceeded the California flood of 1978 led many thinkers to great heights (or depths) in discussing water. One of these calculators supplied some useful data as a tool for cognitation during both events. He had initially calculated to reassure drought victims how much water was left in very dry reservoirs. As we watched hundreds of miles of land with water standing on the surface we re-applied the same calculations:

Given: There are 7.481 gals/ft^3 in a 100 square mile area (10 miles x 10 miles). With water standing only 1/32" deep there are

$$5280 \times 10 \times 5280 \times 10 \times \left(\frac{\frac{1}{32}}{12}\right) = 7{,}260{,}000 \text{ ft}^3 \text{ of water, or } 7{,}260{,}000 \times 7.481 = 54{,}312{,}060 \text{ gals}$$

Just before he went down for the third time an astute government official was heard to declare, "I don't want to be premature but it's just possible the drought is over."

"THINK OF THIS THE NEXT TIME YOU WRITE A LETTER TO THE EDITOR"

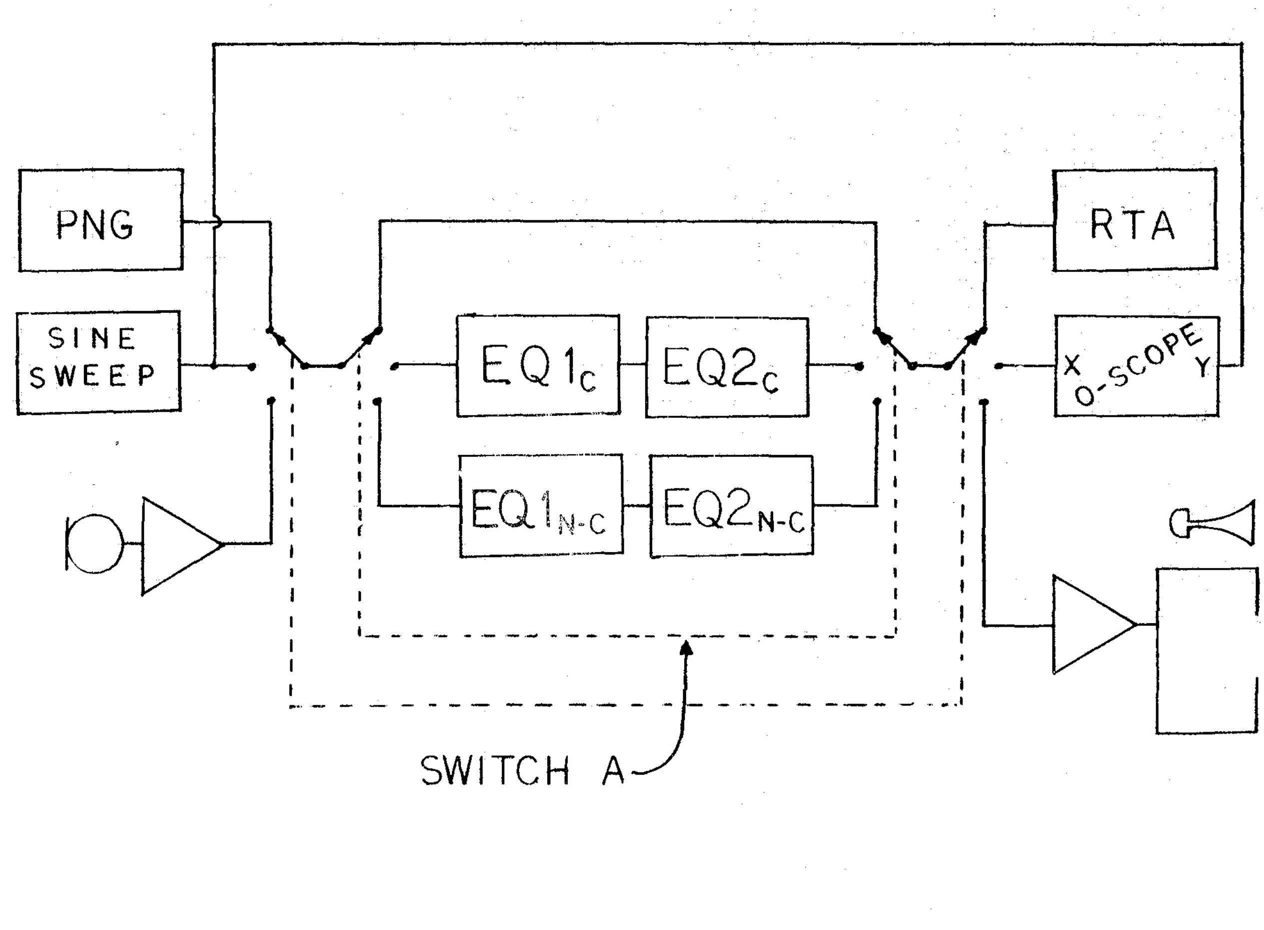
FROM TED UZZLE, CAMBRIDGE, MASS.:

Your exchange of correspondence with Modern Recording (which I do not take) put me to mind of my first letter to the editor. One of those magazines published an equipment review of the Burwen labs hi-fi-type dynamic noise filter. This must be four or five years ago now. Just a few weeks before that I had spent the morning with Dick Burwen in his basement workshop (in which I found the biggest, most beautiful DuMont oscilloscope!). Thus I read the review with particular care. The article said that the input and output levels were 0 dBm, or about three-quarters of a volt. Not having been born yesterday, I flipped to the specs column and found in and out impedances to be around 15,000 ohms, as is typical for hi-fi equipment.

I wrote an ever-so-gentle letter suggesting that the input and output levels were more like -12 dBm.

Eventually there came back a mimeo'd sheet informing me that I could understand decibels if only I'd apply myself. On the back was a log-times-ten table. My next effort at a letter to the editor was a terse opus which Physics Today will be printing sometime this spring.

Shall we co-author an article on the deficiencies of hi-fi-type room equalizers? Here's a demonstration you might want to set up if challenged to show the difference visually and aurally.



TED UZZLE
Consultant
Post Office Box 718
CAMBRIDGE, MA 02139
(617) 492-5585

Carolyn Davis
SHEET NO. six OF
CALCULATED BY DATE 23 March 1978
CHECKED BY DATE
SCALE

Take a cascaded pair of combining octave-band equalizers (I use Shure M-610s), a cascaded pair of non-combining octave-band equalizers (I use the left and right channels of a Soundcraftsmen unit), and a straight wire. Set up a switch for a-b-c tests, as in the diagram. For the straight wire, the combining equalizers, and the non-combining equalizers, perform these tests:

Draw a random, hypothetical house curve on 1/3 octave paper. Or let Punch or Judy do it, some suitable extremity dipped in ink. Setting EQ2 to zero effect, try to match this curve as closely as you can with EQ1, using pink noise and a 1/3 octave real-time. Note how closely you can or cannot match the hypothetical curve.

Adjust EQ2 to correct this same curve, for a flat electrical response. Notice the degree of flatness you are or are not able to achieve, analyzed in 1/3 octaves.

Take a slow sweep sine wave (I use the 30 second sweep on my Wavetek 30) and look at the electrical response on a meter or plotter.

Take the same sweep through the equalizer and look at the phase response of this electrically flat system, with a phasemeter, or an x-y 'scope. Talk through the system, play music through it, etc.

You will appreciate, I'm sure, the purpose and predict the result of each part of the demonstration. It's modest (of time and money) investment, and it drives home to the eyes and ears a point presently left undemonstrated.

TED UZZLE, continued.

Recently I caught up with Florman's The Existential Pleasures of Engineering. I was disappointed, the disappointment of expectation too quickly aroused. One point he could have, but did not, discuss, is the arrogance of the engineer. Now, when in discussion of politics, sports, race, religion, detective fiction, music, sex, or other hobbies, I pride myself on good natured egalitarianism. In engineering, however, I cannot bring myself to believe that all opinions are automatically of equal merit. The abbot Durant had elaborate notions why God's finger held up broken arches. The anonymous masons who actually made gothic vaults stay up must have been idly amused. During China's cultural revolution the epigrammatic thoughts of Chairman Mao were quoted to settle disputes on how best to lubricate factory machinery.

When offered correction based on great knowledge and long experience, I pay close attention. Sometimes I offer it to those who appear not have done their homework. But for those who pontificate ex-cathedra untroubled by knowledge and innocent of understanding, I have no patience. This is surely a failure; it is equally surely a failure to suffer fools.

Place this in the context of the kind of anti-intellectualism we have seen before (Kevles, "Physicists and the revolt against science in the 1930's (sic)", Physics Today, 31:2, pp 23-30, February 1978, and "Depression Does Not Justify Research Work Moratorium", Science News Letter, v. 20, p 397, 19 December 1931), and consider today, a sad age that sees renewed popularity for astrology and the choking off of funds for space exploration; that sees biorhythm calculation a million-dollar industry but genetic research shut off; that finds ancient astronauts believable but Darwinian evolution under the heaviest fire since the Scopes trial; the Bermuda Triangle...Nuclear power.... ours is not a happy time for real science.

How many of Queen Elizabeth's loyal tars sailed against the Armada? In what year? You don't know offhand? Neither do I. But we do know Polonious' injunction; we both know Jacques' meditation on the stages of life; we both remember Hamlet's words on looking death in the face.

Which of the Borgias slept together? Which poisoned each other? You don't know off hand? Neither do I. But we both know "The Last Supper", with all those shocked faces, each looking within himself with horror. We each shudder on remembering Michelangelo dashed a sculpture to bits when a wealthy patron quarreled over the price.

Few, it would seem, have appreciated the contemporary accomplishments that have let each age live in the collective memory. We create greatness casually, toss it aside as a triviality, and turn to some grim, bloody business: some hate, anger or fear whose ashes will go cold overnight. What has the human race gained permanently from our time, putting granola-munchers on Vermont communes or putting astronauts on the moon?

Consider this: is democracy served by a bridge king and commoner alike may use to cross the river? ...given the unequal benefit each will derive from getting to the other side?

Consider this: is democracy served by a bridge that collapses under the feet of king and commoner alike?

As you know, Leo Beranek is president of one of the local television stations here. He may be our modern St. Francis, able to talk with the birds and flowers and others who have not mastered the differential calculus. Once a month they have a program during which viewers may call in and ask questions of the top management. Once a month someone calls in to ask, why are the commericals so much louder than the programs? - and once a month Beranek commits an act of obscene understatement: he says, "I know something about that." This is followed by an elegant dissertation on peak vs. program levels, compansion, carrier modulation, and similar arcana.

Think on this the next time you write a letter to the editor.

Yours very truly,

Ted Uzzle

THE CROWN EQ-2 EQUALIZER DOES COMBINE

On Page 16 of the last Newsletter, Vol 5 No. 2, Norman Eisenberg's answer to Carolyn's mis-guided Letter to the Editor is reproduced. Mr. Eisenberg casts asperions on the Crown EQ-2 by including it along with some questionable equalizers with his statement, "If we are to believe the writer of this letter (Carolyn), then in addition to Delta-Graph, such companies as Crown, ... et al also do not understand equalization. This seems to us a rather untenable and unprovable position."

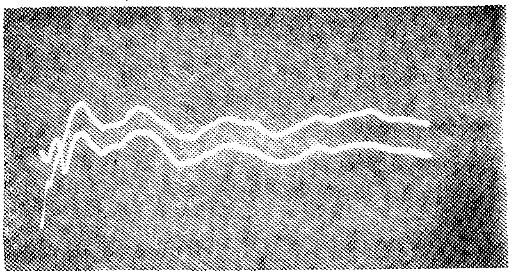
The Crown unit *ABSOLUTELY* is a combining-type filter (and as smooth a combining filter as we have had the pleasure to measure). To test these excellent, carefully engineered filters, tune any two of them until their "skirts" cross over at their 'half pad loss points'. They will then provide a 6 dB dip between the center frequencies of the two if each one is set to -3 dB.

During the San Francisco class we demonstrated this to the class because one of the members of the class had been told that the Crown EQ-2 did not combine. The UREI model 200 level recorder quickly revealed that *it did indeed combine and beautifully.*

It seems a shame that so many reviews on equalizers are *totally* worthless because the testers lack a knowledge of the fundamentals.

GAIN INHERENT IN FLUSH MOUNTING MICROPHONES

Time delay spectrometry (TDS)allows anechoic measurements to be made with ease in non-anechoic environments. Here are two response measurements of a microphone flush with a surface (top curve) and the usual table mike stand distance above the surface (bottom curve), showing the gain available from flush mounting.



They are exactly 6 dB apart over the whole range (approximately 1,000 Hz to 10,000 Hz.)

We felt this picture illustrates in even greater detail than the previously published 1/3-octave curves the effect of the coherent summing that occurs in the "pressure field" right at the surface itself.

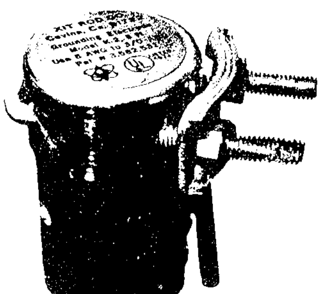
"A NEGATIVE APPROACH TO GROUNDING"

HAROLD LINDSAY sent us the data sheets reproduced below along with his recommendation for the XIT Grounding Rod. Increasingly the sound contractor finds that he must take the responsibility for finding his own ground rather than using the traditional approaches. The XIT rod should solve many, many problems.

XIT ROD

Listed 467J

A CHEMICALLY-CHARGED-ROD ELECTRODE



This is the first and only electrolytic grounding rod to obtain Acceptance for Underwriters' Laboratories Listing granted under UL467J. The Electrical Council of Underwriters' Laboratories Inc. adopted Revisions of the Standard for Grounding and Bonding Equipment, in Appendix A, paragraphs 6.7L, 6.7M, and 6.7N exclusively covering the patented Xit Rod system. Approval as an American National Standard has been established under ANSI-C33.8. Electrolytic grounding has now come of age with Xit Rod, which derives its operating moisture from the atmosphere forced into the interior chemicals by the gentle pumping action associated with barometric pressure changes.

SPECIAL FEATURES

Grounding in earth is variable by (1) soil type, (2) chemical content, (3) moisture, and since the size and depth of Xit Rod is compatible for good grounding, it also overcomes the three variables above by generating its own moisture and metallic salts and, most importantly, uses the resulting electrolyte as the interface medium between the conducting surface of Xit Rod and outwardly in the soil, which lowers resistance markedly.

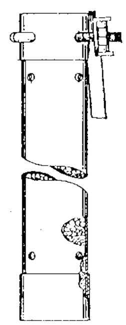
Its conception and development results from the impasse between building, plumbing, and electrical officials and their codes because of:

(1) The expanding use of plastic pipe which excludes its use for grounding.
(2) The costly degradation of galvanic action in metal systems and equipment.
(3) Shock hazards of water pipe grounding.
(4) Because earth grounding is so variable and difficult.
(5) Safety devices require consistent low resistance to ground.
(6) Freezing temperature has no effect on Xit-Rod or its function.
(7) Xit-Rod functions indoors or out.
(8) Brazing directly to Xit-Rod is not harmful, although it may dry out and cause a short delay in elecrolyte formation in that portion of rod heated.

"TRULY THE MOST NEGATIVE APPROACH TO GROUNDING"

TECHNICAL SPECIFICATIONS

Xit-Rods are used in any electric, communication, or process system, in arid or wet conditions, and their impedance to ground averages 5 ohms or less. They cost less to use than any other made or investment system because only one Xit Rod is necessary as compared to the plurality of other rods usually required. No periodic checking or adjustments are necessary.



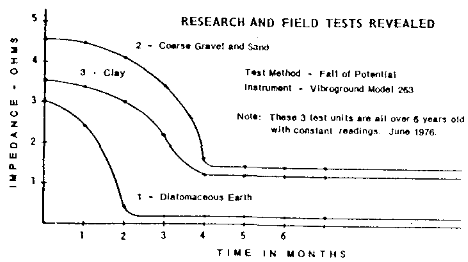
Diameter of all models is 2-1/8" O.D. - - Wall thickness - .083"
Material - Type K Copper - - Galv. Steel wall - .110

.K2-8 Standard model for general use 8' lengths.
.K2-10 Is recommended for installations in jurisdictions where 10' depth is minimum requirement, and in areas where a high resistivity factor is known.
.S2-10 Of dip galvenized steel for use in systems where intermettallic or galvanic reactions must be avoided, for example, anchor points for guy wires and towers.
.K2-H Where bedrock is encountered too impervious for drilling with auger bit. Use this L shape horizontal model. Specify vertical depth requirement.
.K2-20 Is available in 20' length for power stations, microwave and communication towers, where there exists known earth resistance variables too great for good consistent communication. Also in areas where installations are vulnerable to lightning.

*Note: Resonant lengths for RF frequencies on special order.
Other lengths available - write needs
Ground Clamp For - #8 AWG to #3/0 AWG
Cable Fastener. U-Bolt and Pressure Plate furnished
ALL U.L. APPROVED. "Our best to your Electrical Systems."

* Listed 467J

Cordially, XIT ROD
GROUNDING DEVICES
P.O. Box 128 — Beaumont, CA 92223-- (714) 845-3988



Patented U.S. #3,582,531, U.K., Canada, Europe, Near East, Asia and So. America

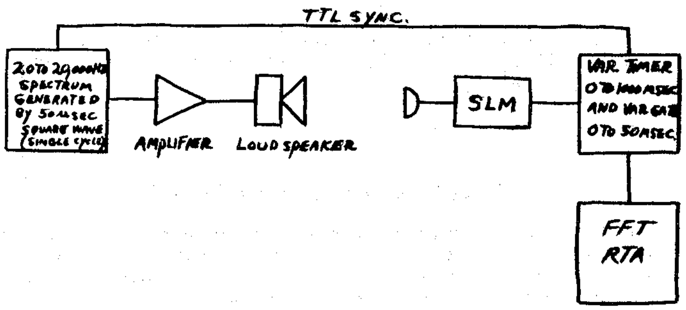
The XIT Electrolytic Ground Rod lists at $159 for 8 ft.; $189 for 10 ft; $199 for 12 ft; and $359 for 20 ft. Their address is XIT Rod Co., P O Box 128 Beaumont, CA 92223 (714)845-3986.

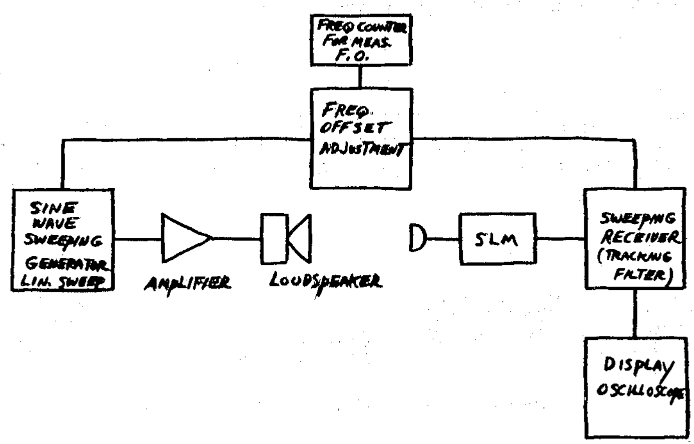
CURRENT STATUS OF T.D.S. WORK

There has been substantial experimental work on finding ways to adapt, modify and convert existing analyzers into Time Delay Spectrometry (TDS) instruments. Current endeavors include:

1. FFT-TDS
2. Linear sweep - frequency offset
3. Linear sweep - time delayed
4. Interrupted pulse technique
5. Tracking filter technique

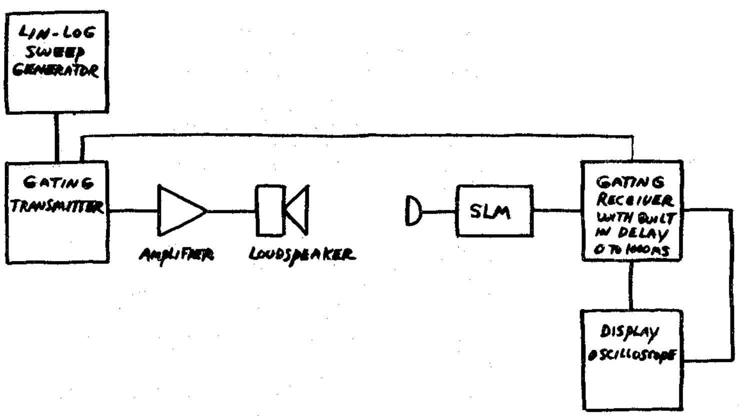
The single line block diagrams show the experimental set ups. In addition to the diagrams we have included the equations necessary to actively pursue TDS work accurately. This is not to say that there are not many other possible techniques. These are the ones currently under inspection.

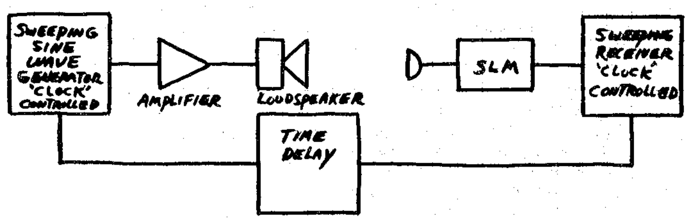




FFT-TDS, excellent method - allows both log and linear display. High resolution (400 lines). Costly (over $11,000). Remarkably versatile spectrum analyzer

Linear Sweep - Frequency Offset Technique. The original method developed by Dick Heyser and used the past five years by Dick Heyser and Cecil Cable. Costly (around $12,000). Difficult to maintain and calibrate. Has allowed pioneering work to be done. Versatile general use spectrum analyzer in addition to TDS.

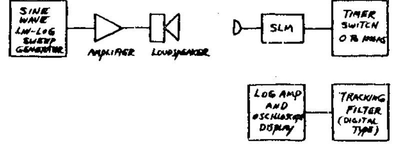




Interrupted Pulse Technique - May be least costly of present techniques ($3200 for gating device) slower to use in reverberant spaces.

Linear Sweep - Time Delayed - Can be done with existing equipment. Display can be either linear or log. Cost is approximately $5000 without Time delay. Useful spectrum analyzer for general use.

Tracking Filter Technique - have not found a suitable tracking filter as yet - difficult to calibrate. Tracking filter cost approximately $3,500



CURRENT STATUS OF TDS WORK, continued

SWEEP FREQUENCY EQUATIONS

Definitions:

ST is the time in seconds for a sweep from a LFL to a UFL

BP is the bandpass in Hz from a LFL_F TO A UFL_F

BP_R is the bandpass ratio $\left(\frac{UFL_F}{LFL_F}\right)$

SR is the number of cycles swept in one second

%BP is the percentage bandwidth

BPST is the time in seconds that it takes to sweep from LFL_F to UFL_F

UFL is the upper frequency limit of sweep

LFL is the lower frequency limit of sweep

UFL_F is the upper frequency limit of a filter

LFL_F is the lower frequency limit of a filter

N_F is the number of filters that can be placed contiguously between the LFL and the UFL

f/oct is the fraction of an octave

f_c is the center frequency a filter is tuned to

LINEAR EQUATIONS	LOGARITHMIC EQUATIONS
1. $ST = \frac{(UFL - LFL)}{SR} = \frac{BPST(UFL-LFL)}{BP}$	$ST = \frac{\ln\left(\frac{UFL}{LFL}\right)(BPST)}{\ln BP_R}$
2. $SR = \frac{(UFL - LFL)}{ST}$	$SR = \frac{UFL - LFL}{ST}$
3. $\%BP = \left(\frac{BP}{f_c}\right)*100$ *f_c is the center frequency	$\%BP = \left(\frac{BP}{f_c}\right)100$
4. $BPST = \frac{ST(BP)}{(UFL-LFL)}$	$BPST\ \frac{ST(\ln BP_R)}{\ln\left(\frac{UFL}{LFL}\right)}$
5. $(UFL - LFL) = SR(ST)$	$\left(\frac{UFL}{LFL}\right) = e^{\left(\frac{ST(\ln BP_R)}{BPST}\right)}$
6. $N_F = \left(\frac{(UFL-LFL)}{BP}\right) + 1$	$N_F = \left(\frac{\ln\left(\frac{UFL}{LFL}\right)}{\ln BP_R}\right) + 2$
7. $f/oct.** = \frac{\ln\left(1+\frac{\%BP}{100}\right)}{\ln 2}$ $** \frac{1}{f/oct.} = \frac{f}{1}$	$f/oct. = \frac{\ln BP_R}{\ln 2}$

Notes on BP_R

$$BP_R = \frac{UFL_R}{LFL_R} = 1 + \frac{\%BP}{100} = \frac{BP \text{ in Hz}}{f_c} + 1 = 2^{f/oct.}$$

Notes on Linear Sweep TDS

$$\text{Freq. offset (FO) in Hz} = SR\left(\frac{\text{Distance}}{\text{Velocity of Sound}}\right) \qquad \text{Distance} = \frac{(\text{Velocity of Sound})}{SR} FO$$

Some Selected Examples

1. Linear sweep from 100 to 10,000 Hz in

   1 sec $\left(SR = \frac{UFL - LFL}{ST} = \frac{10{,}000 - 100}{1} = 9{,}900 \text{ Hz/sec}\right)$

   If I use a filter with a BP = 10 Hz, how many milliseconds does it take to sweep 10 Hz?

   $$BPST = \frac{ST(BP)}{(UFL - LFL)} = \frac{1(10)}{9900} = 1.01 \times 10^{-3} \text{ seconds or } 1{,}000 \times 1.01 \times 10^{-3} = 1.01 \text{ msec}$$

2. Logarithmic sweep from 100 to 10,000 Hz in .25 sec (ST = .25 sec). SR still is 9,900 Hz/sec. If I use a 1/3-octave filter set how many milliseconds will the sweep take to pass through the BP of one filter?

   $$BPST = \frac{ST(\ln BP_R}{\ln\left(\frac{UFL}{LFL}\right)} = \frac{.25\left(\ln\left(\frac{400}{315}\right)\right)}{\ln\left(\frac{10{,}000}{100}\right)} = 13 \times 10^{-3} \text{ or } 1{,}000 \times 13 \times 10^{-3} = 13 \text{ msec.}$$

   Since .25 secs = 250 msec and the sweep will spend 13 msec on the first and 13 msec on the last filter and $\frac{250}{13}$ number of equal intervals in between the first and last filters we can assume that ≈21 filters are involved.

CURRENT STATUS OF TDS WORK, continued

Using the equation for N we find

$$N_F = \left\{\frac{\ln\left(\frac{UFL}{LFL}\right)}{\ln BP_R}\right\} + 2 = \left\{\frac{\ln\left(\frac{10{,}000}{100}\right)}{\ln\left(\frac{400}{315}\right)}\right\} + 2 = 21.3 \text{ filters}$$

3. What would have been the nubmer of filters between these two frequencies above (including the first and last frequencies) if we had used linear sweep and a filter with a BP = 10 Hz?

$$N_F = \frac{(UFL - LFL)}{BP} + 1 = \left(\frac{10{,}000 - 100}{10}\right) + 1 = 991 \text{ filters}$$

4. Suppose that you wish to sweep a 23% filter so that the sweep takes 10 msec to pass through each 23% BP and do this with a ST = .25 secs. For a starting frequency of 50 Hz what UFL would apply?

$$\frac{UFL}{LFL} = e^{\left\{\frac{ST(\ln BP_R)}{BPST}\right\}} = e^{\left\{\frac{.25\left(\ln\left(\frac{400}{315}\right)\right)}{.01}\right\}} = 392.41 \qquad UFL = LFL \times 392.41 = 19{,}620 \text{ Hz}$$

5. I am using a linear sweep and would like to select a SR that would insure that my filter's BP = 10 Hz was swept in 5 msec (BPST = 5 msec). If I am sweeping from 0 to 10,000 Hz what should my SR be?

$$ST = \frac{BPST(UFL - LFL)}{BP} \quad \text{and} \quad SR = \frac{UFL - LFL}{ST}$$

thus

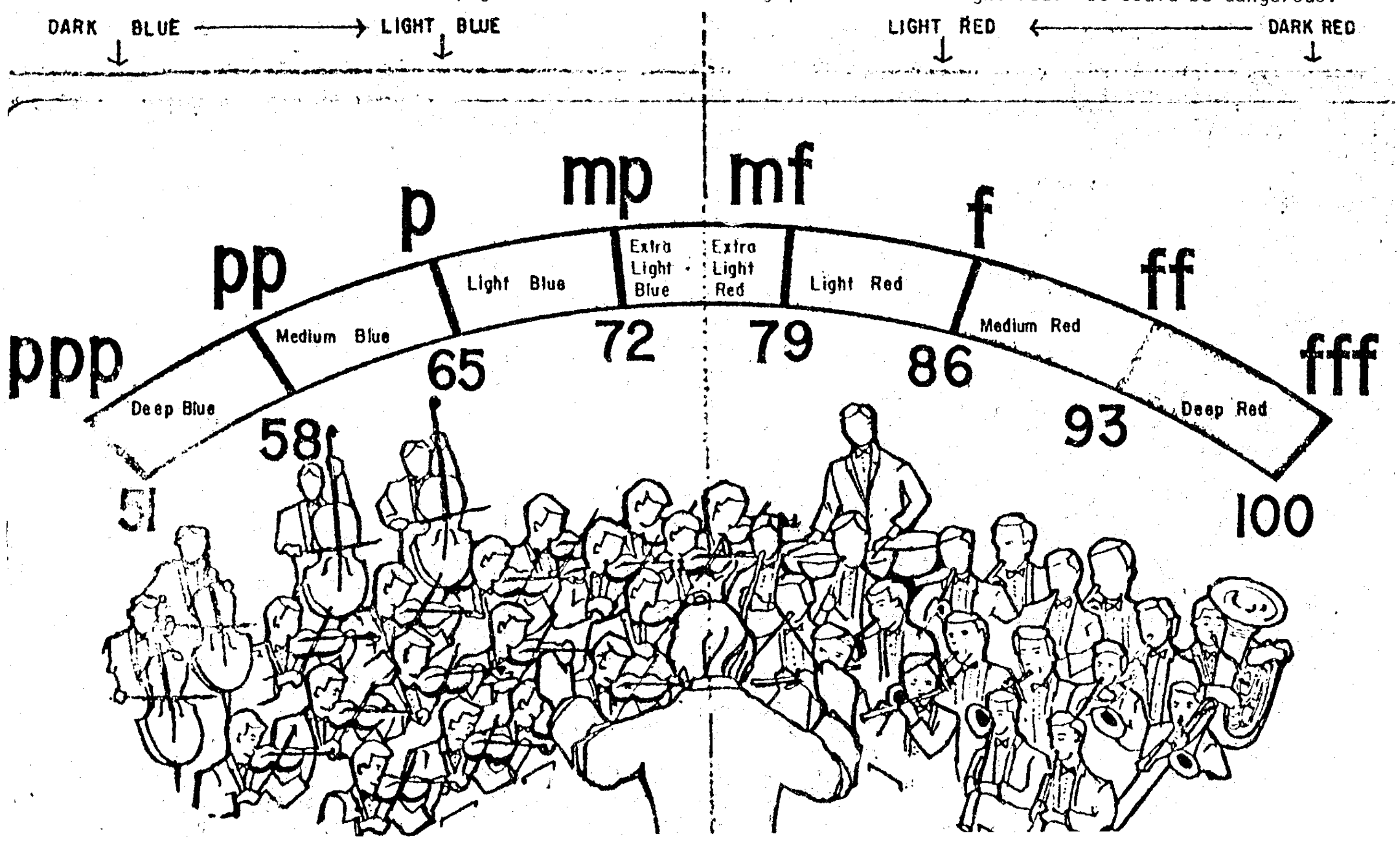
$$\frac{.005(10{,}000)}{10} = 5 \text{ secs} \qquad \text{and} \qquad \frac{10{,}000}{5} = 2{,}000 \text{ Hz/sec}$$

Manipulation of these equations, such as pulling ST out of Eq. 4 as was done in the last example, can provide interesting and worthwhile insights into the uses, limitations, and advantages of both linear and logarithmic sweep techniques.

DBD

INTERESTING SPECIFICATION

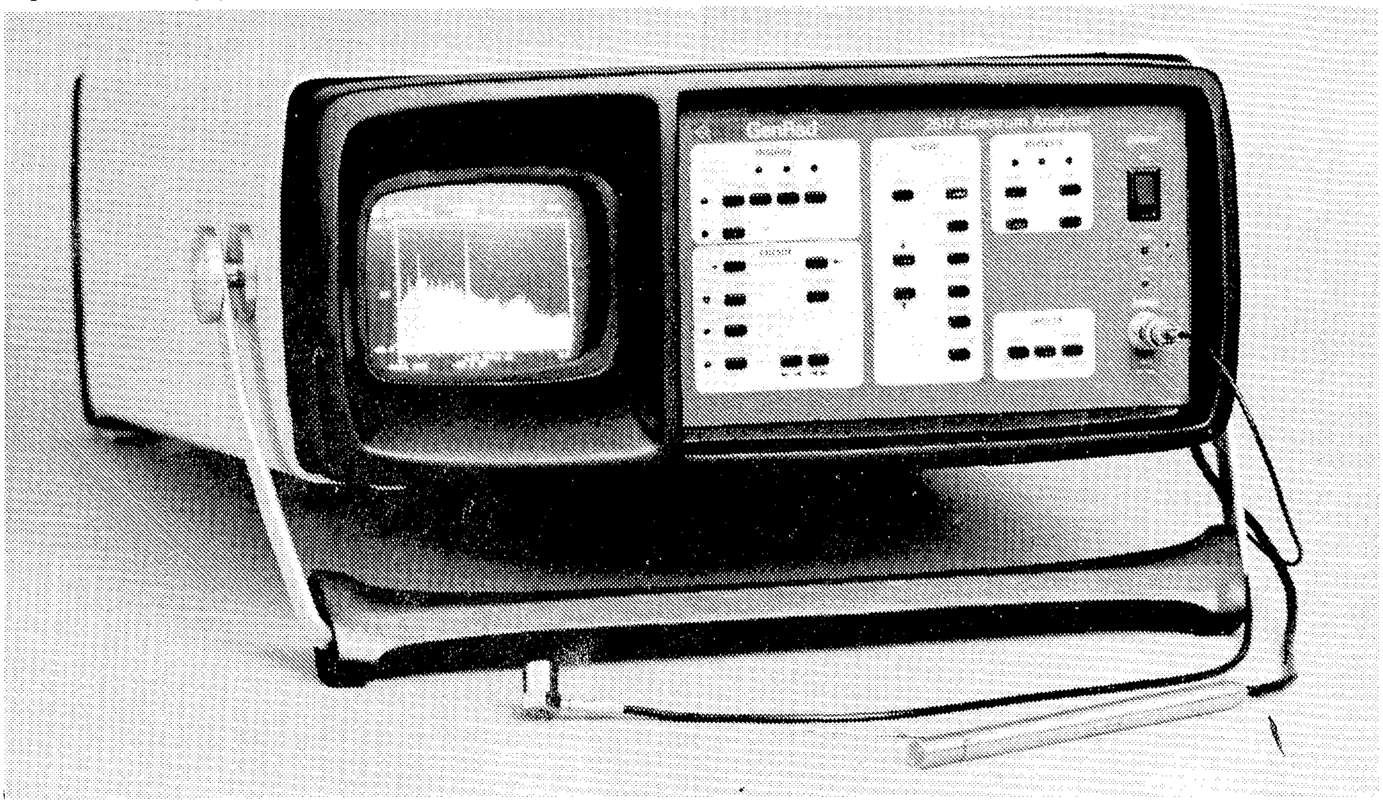
TERRY HOFFMANN, Johnson Controls and a grad of the Chicago 1977 class, knows that we are eager to see sound system specifications, especially anything new and interesting. He sent us one along with a few comments, "The operator's console and music level indicator are two of the features which are most unusual. While the console appears to be a triumph in terms of human engineering and esthetics,........The music level indicator intrigues me for a different sort of reason. With this type of unit being specified, how long will it be before some over-eager sound man informs a Fiedler or Bernstein that the fff on page three of the score only peaked in the light red. It could be dangerous!"



FFT SPECTRUM ANALYZERS

Fast Fourier Transform(FFT) analyzers have been around for a number of years at very high prices and in very large packages. Primarily academia's plaything or part of a federally funded test (fft) project, the rest of us could only speculate as to their usefulness in our own measurements.

We now know of at least two instruments that offer the desired capabilities within budget figures, albeit at the high end, of many professional audio engineers.



The first of these new units (see above) is the GenRad 2512 spectrum analyzer. We have already had a chance (one evening after the San Francisco class) to use this unit in a protoype form.

Using a 50 microsecond (u sec) tone burst (one cycle of a 20,000 Hz/sec square wave) fed directly into the analyzer from the generator gave a 20-20,000 Hz flat spectrum. This same signal fed via the power amplifier to the loud-speaker and picked up by the SLM and fed to the analyzer gave a full frequency response reading each time the short pulse was triggered.

The raster-scan CRT allowed extremely stable alpha-numeric displays that are calibrated in plain language text rather than symbols. All control settings are made by looking at "menus" on the CRT and using a cursor to choose the parameters desired. When a "menu" choice is made the screen then calibrates itself to the choice. Impossible combinations are automatically locked out as the choices proceed. One picture of the CRT includes the curves taken plus all control settings chosen listed on the screen. The memories can restore both the curves and the calibration info even after the analyzer is turned off and then on again at a later time.

The instantaneous as well as averaged spectrum is fully calibrated in the units chosen by the operator. In addition to the standard logarithmic dB display, linear amplitude (volts), and power (volts squared) are selectable to achieve results consistent for sinusoidal signals. Calibration in power density consistent for random signals and energy density useful in transient analyzers is provided without the need for manual conversion. The operator may also enter his own calibration in dB or millivolts. The frequency axis can be displayed in Hz or CPN, and either linear or logarithmic scales are available.

Averaging modes include additive, subtractive, exponential, and maximum hold.

With 400 line storage, 1MV sensitivity full scale, and a weight of just under 40 lbs, it's fair to say progress is being made. This unit should be available for shipment sometime in the May-June period. Price approximately $11,000.

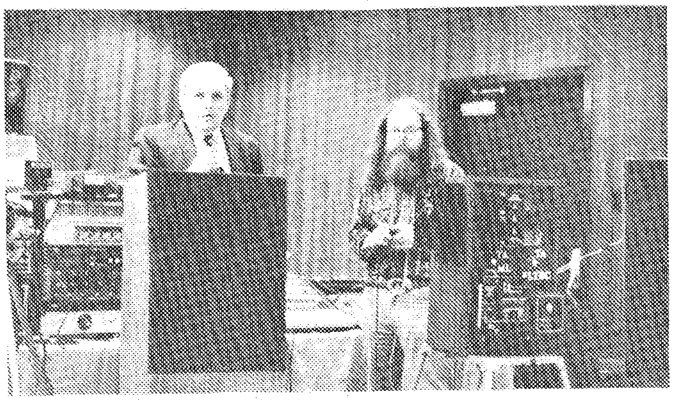
The second unit, just announced, is from Hewlett Packard. We have not seen a prototype but have received a zerox of an internal spec sheet from our friendly HP contact. The model 3582A spectrum analyzer is a two channel FFT. It also will sell for approximately $11,000.

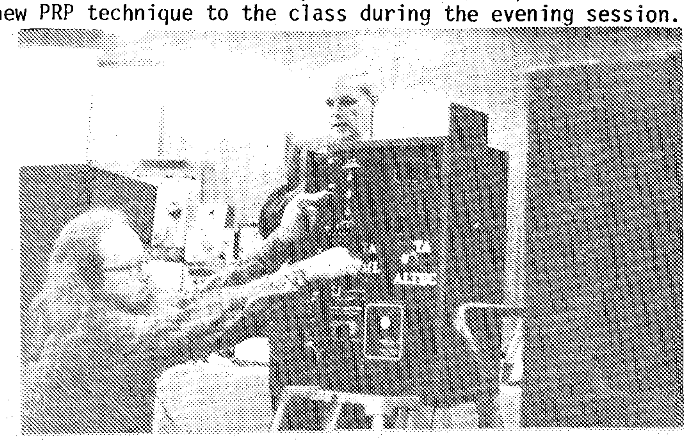
What all this means is that Syn-Aud-Con will more than likely be doing FFT-TDS and FFT phase spectrum of loudspeakers in the Fall-Winter classes of 1978-1979. We hope to have definite information on this in the July Newsletter.

DBD

THE PRP MICROPHONE TECHNIQUE

At the San Francisco class in February this year, ED LONG and RON WICKERSHAM, easily two of the most productive independent experimenters in audio today, revealed their new PRP technique to the class during the evening session.





PRP stands for "pressure response pickup" and it is a truly intelligent application of fundamental acoustic "laws" hitherto not applied properly with regard to microphone technique, and yet instantly obvious that it should have been done this way all along.

The technique consists of orienting the microphone vertically above the floor (a wall or ceiling, if hard surfaced, would do as well) and adjusting it so that its diaphragm is as close to the floor surface as possible without actually touching.

Time Delay Spectrometry measurements have shown all of us that it's not possible to get too close to a surface. So far this smacks of "flush mounting" and indeed it is a variation of it. In this case, however, the floor and diaphragm are forming a cavity with a top (the diaphragm) and a bottom (the floor) and a cylindrical wall (the viscosity of the air).

Herein lies the first surprise -- the microphone needs to be pressure response calibration. (A pressure field is one in which the instantaneous pressure is everywhere uniform. There is no direction of propagation. The pressure field exists primarily in cavities, commonly called couplers, where the maximum dimension of the cavity is less than 1/6 of the wavelength of the sound. See Newsletter Vol 5, # 2, page 10 for a discussion of Sound Fields).

How obvious but how unheeded by us all in the past. Ed and Ron use specially modified 1/2" B&K measurement microphones. (The Knowles Electronics 1759T miniature microphone shown in class *is a pressure response microphone.*)

Placement of the Microphone

Ed and Ron use pressure response microphones to make what we believe are the finest stereo recordings we have heard. So far they have recorded small groups (spacing width of group about 14 to 16 feet). Their method is to space the microphones outside of and to the front of the group using a spacing distance of *20 to 25 feet.* Think about the significance of that distance for a moment. Twenty-five times .885 msec/ft ≈ 22 msec. That means that a signal originating near one microphone but loud enough to be "heard" by the other microphone will not be detected by a listener because of the "Haas Effect" provided by the 22 ms delay *between* the two microphones. Again, ridiculously obvious but, so far as I know, never consciously applied as a microphone pickup technique.

Ed and Ron space their playback T.A. monitors about the same distance as the width of the group.

Subjective Reaction

The playback of stereophonic material possessed the finest spatial localization I have heard from a two channel system. One listener remarked "I felt as if I could *walk around* the singer." Great freedom in listening position while retaining a locked-in geometry across a curtain of sound was evident.

It is necessary to place the loudspeakers well out away from the nearest reflecting surfaces in order to avoid wide-band anomolies (about 10' from the nearest wall). This would be accomplished in a control room with extensive absorption around the loudspeakers.

PRP provides significantly superior recording of amplitude and time-phase relationships with the resultant improved stereo geometry and localization. The sound, free of broadband anomolies, is measurably and audibly smoother and cleaner. Ed and Ron are to be congratulated on achieving a genuine breakthrough in an art that is older than they are.

DBD

KNOWLES SUBMINIATURE MICROPHONE

As we mentioned above, since writing the announcement in Newsletter Vol 5, No. 2 that we have ordered the Knowles subminiature microphone (the 1759T) to have available for Syn-Aud-Con graduates in small quantities, we have found out that it is a pressure response microphone. The write-up above on PRP describes the importance this has. We also ordered the Knowles CB 1846 miniature speaker (about 1"x1"x¼"). You may order it from us for $15 each.

cd

ARTICLES IN dB MAGAZINE BY HAROLD LINDSAY

Anyone who has experienced a historical event as a participant or as an eye witness knows all too well how distorted that event can become when described by others who were not present but are working from written documentation, hearsay, and interviews of uninformed bystanders.

History of all sorts and particularly history of the audio industry has been of lifelong interest to me. It has been my privilege to meet, know and enjoy friendship with many of our industry's most talented contributors. Few among them have had the impact on our audio experience to equal that of HAROLD LINDSAY who was literally the engineer behind the foundation and success of the Ampex Corp.

Harold's remarkable talents have led him to be one of the founders of Emilar, a successful sound contractor and a sought-after audio consultant since retiring from Ampex.

Recently he has revealed still another of his multitude of talents by writing a readable and reliable history of the beginnings of magnetic recording here in the United States. (dB Magazine, December 1977 and January 1978.)

That real drama can and does occur in an audio engineer's life is evident in one passage of Harold's article. After designing the playback heads, they were to be compared to Jack Mullin's Magnetophon heads. Harold writes:

> I have always remembered that next moment, just before pressing the start button, as one of the most anxious times in my entire life - so much hung in the balance; a dismal failure or the beginning of an exciting future. The tape whipped up to speed; we were stunned, entranced, suspended in an eternity of mere seconds. Then wild celebration. Our ears had just told us what measurements later confirmed - we had outperformed the Magnetophon head. We were destined not to failure, but to fame.

Shakespeare wrote:

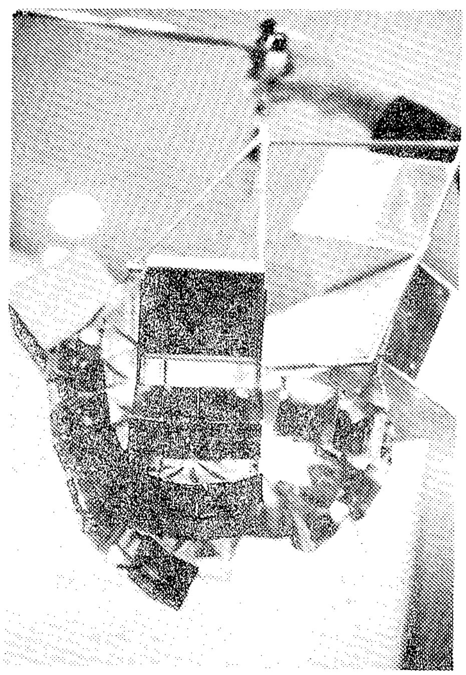
> There is a tide in the affairs of men,
> Which taken at the flood, leads on to fortune;
> Omitted, all the voyage of their life
> is bound in shallows and miseries.

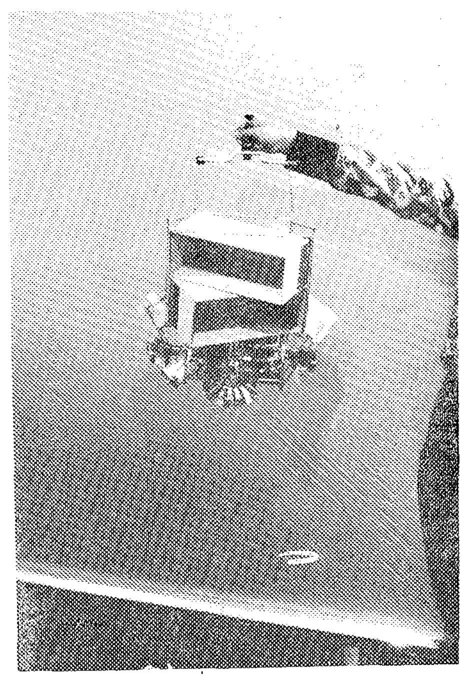
Harold Lindsay's two articles, the first in the December 1977 issue of dB Magazine, pages 38-44 and the second in January 1978, pages 40-44 tell the tale of men soaring on the flood crest. That he catches the excitement, the challenge and the adrenal-pumping hopes and fears in addition to lending history a hand with the "straight story" makes these two articles collector's items, in my judgment.

DBD

MODEL ARRAY BUILT BY BOB BECKER

In discussing arrays a picture easily is worth a thousand words and a model - well let's let the illustrations tell the story.





BOB BECKER, of Advanced Sound in Sacramento, and a 4-time graduate of San Francisco classes, wanted to know what it was going to look like when he studied the drawings of an array proposed for one of his jobs. He built it in miniature. The San Francisco class was impressed with the stunning detail Bob provided. Note in the close up that each cell in the multicellular horns is independent.

No wonder Bob looks pleased. He's impressed the customer, the consultant and his peers.

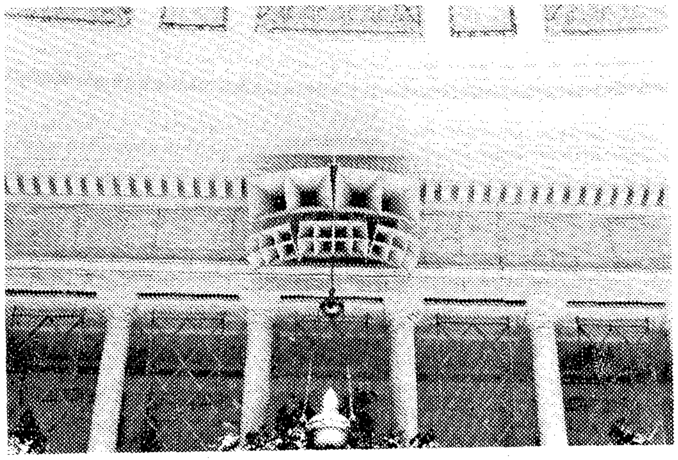
DBD

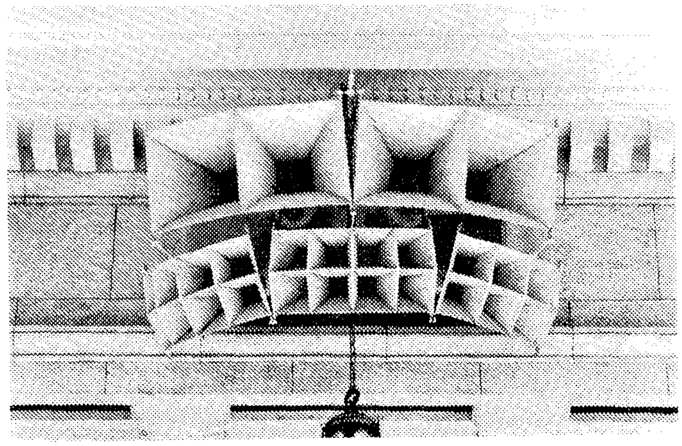
TEMPLE ISRAEL SYNAGOGUE

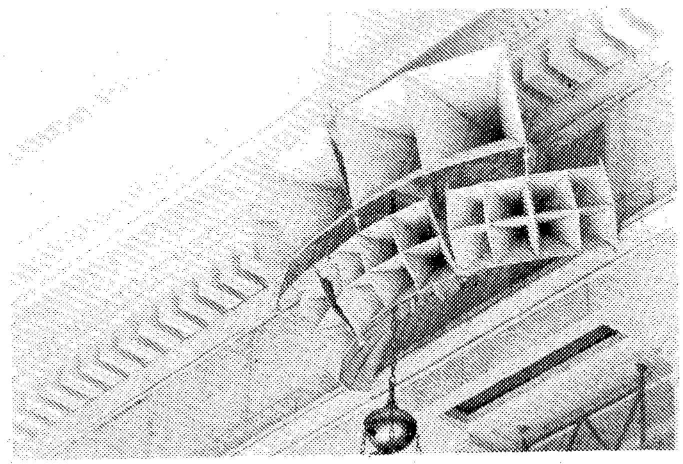
From ED LETHERT, consultant in Minneapolis: Early in 1977, Temple Israel began the complete remodeling of their main sanctuary.

Northwest Sound was called in to upgrade the existing sound system, part of which was the addition of three horns to the existing array of two. This was surely going to make the array more obvious and compromises in sound quality were made (side-by-side arrangement of horns) so that the appearance of the horns was natural and uncluttered.

The first step was to flush mount the bass speakers in the wall behind the array. The horns and drivers were then painted and glazed to match the finish of the walls. The rear of each unit, where most of the weight is concentrated, was securely fastened to the iron structural members with heavy chain. Minimal support was required in front, the primary need being to hold each horn in its proper position.







The photos show the result of conservative use of rod and fasteners to accomplish this. From the floor, the mounting hardward is hardly visible, and according to Temple staff members, the array is completely acceptable and presents a pleasing appearance.

"360° SPACE IN TWO CHANNELS"

Peter Scheiber, as dB Magazine says, can rightly be called the "father" of four channel matrix. He has written an interesting article in the January issue of dB called *360° Space in Two Channels*. He ends his article,

"Who needs four channels? I've heard some great spatial sound reproduced via four discrete channels; my own preference has been to investigate the potential inherent in phase-amplitude coding 360 degrees of auditory space in the two channels that we already have. Through control of phase information, the "stereo" pair of channels can carry not only a flat, left-right wall of space, but a full 360 degrees of surround space. We've been wasting half the spatial information capacity of our pair of channels. Let's put it all to use."

It would seem that the "father of quad" has just given last rites to his offspring.

Cd

THE NEW SHURE V15 TYPE IV PHONO CARTRIDGE

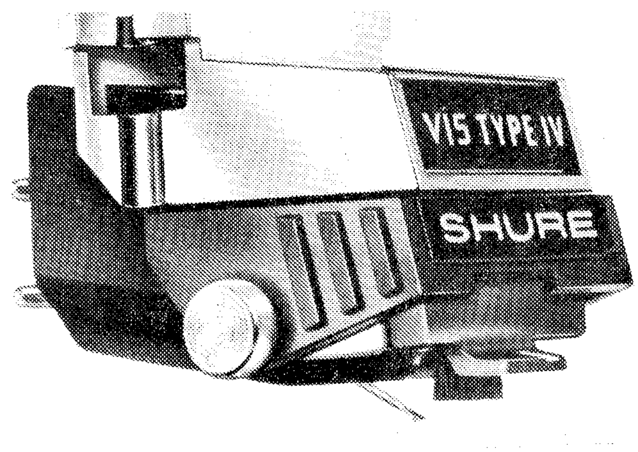
As a happy owner and user of a Shure V15 Type III in an SME arm along with a Technics direct drive turntable, I have resided in phonograph "hog heaven" for the past several years. Therefore, when Shure announced a V15 Type IV my first reaction was one of "What can Shure do to an already outstanding cartridge that can be *audibly* worth a change?"

What they could, and did do was to turn my own guiding engineering philosophy on me by approaching the *entire* problem of getting the signal out of the phonograph disc's groove into the input of my preamp as a *system* problem to be solved in terms of the interaction of all the elements encountered in the path.

Listening Experiences

Syn-Aud-Con doesn't do, and doesn't believe in "Tested in the Home" type reports - unless one has the knowledge and integrity of Dick Heyser and an editor as ethical as Gene Pitts of Audio. What we can report with accuracy and fairness is our *subjective* experience relative to our past experiences with this versus other units and our *opinion* with regard to this specific unit's performance with recorded material we are familar with.

There is a clearly audible difference and an unbelievable improvement in trackability and stability. Records in our personal collection that have suffered damage of one sort or another but are precious to us for the performance are now playable again. One record would almost bounce the SME arm with the Shure V-15 Type III into the air as a result of an "air bubble" bump in its surface. The Shure V-15 Type IV in the same arm tracked the bump easily with only the slightest tic as it passed the bump. An old, worn, noisy recording of a string quartet lost some of its shrillness and became more listenable.



This has been accomplished by a team of extremely talented engineers who have not only accepted the challenge of the "systems approach" but have integrated their special insights into a new standard for an industry.

The accompanying Tech Topic is a reprint of the introduction to a complete manual prepared for the presentation of the cartridge, *Introduction - an Integrated System Design*, by James H. Kogen, Vice President of Engineering at Shure Brothers. His comprehensive introduction plus a selected set of illustrations touching on the most innovative features of this new cartridge are worthy of study for anyone interested in a classic engineering project.

My Technics turntable is mounted *on top of my Emilar speaker system (right channel)*. I can turn the level up until the IOC lights on the Crown amplifier *without any* feedback from the loudspeaker to the cartridge.

REVERBERATION TIME VS DECAY RATE

The reverberation time of an enclosed space is defined as the length of time in seconds that it takes sound to drop 60 dB in level upon having the electrical driving signal to the sound source turned off. This has been symbolized as T, T_{60}, RT_{60}. When no frequency is specified it may usually be assumed that the measurement was made in the 500Hz octave band because of past historical practices.

A useful notation might be

$RT_{60}(500)$ or $RT_{60}(2K)$, etc.

Rarely, if ever, is the full 60 dB of decay observed. Quite often only 15 or 20 dB constitutes the total sample and the times figure is extrapolated to what the 60 dB figure would have been.

A second way of expressing exactly the same data would be to use the decay rate (D) and use the subscript to denote fractional octave bandwidth and center freuqency such as

$D_{1/1(500)}$ or $D_{1/1(2000)}$ (Note from the typist: This idea will never "fly")

where $D = \frac{60dB}{RT_{60}}$ and is in dB/sec

Thus, if we measure in the articulation region with an octave filter and find that the $RT_{60} = 2.5$ secs we would write

$D_{1/1(2000)} = 24$ dB/sec* $\qquad \frac{*60\text{ dB}}{2.5\text{ secs}} = 24$ dB/sec

If two slope rates were involved as can, on occasion occur, then a further description could be

$D_{1/3(2000)15} = 24(20)$ dB/sec

which would mean that the 24 dB/sec rate was valid for only the *first* 15 dB of decay and the remainder of the decay sample was at 20 dB/sec in the 2000 Hz, 1/3 octave band. This type of technical shorthand would be of immense help in making present day measurements more meaningful to latter day workers who might wish to critically evaluate your measurements regarding a space that had survived the intervening period.

THE DAVID CLARK COMPANY WIRELESS HEADSET SYSTEM

The David Clark Co., as every Syn-Aud-Con graduate must know by now, is the manufacturer of Judy's hearing protectors. They have now developed what just might be the best way for wireless paging to be accomplished.

We're publishing their preliminary write-up on this product, not as a new product announcement, but rather as a "market survey" so those of you who have an interest or need of such a product can contact Ron Roscoe at David Clark Company, 360 Franklin St., Worcester, MA 01604. (617) 756-6216.

GENERAL INFORMATION

The David Clark Company FM Induction Receiving System is designed for one-way, wireless, high quality communication. It is especially useful in a high-noise environment, where hearing protection is required.

The system is based on the concept of the transformer: when two coils of wire are placed in close proximity, and an alternating current flows in one coil (primary), a similar alternating current will be *induced* in the other coil (secondary).

In the DCCI FM Induction Receiving System, a single turn primary "loop" encloses the area in which the system is to function. The secondary is a multiple turn coil built into the receiving headset. As long as the headset is within the primary loop, a signal will be induced in the receiving coil. Reception drops off sharply outside of the area enclosed by the loop, preventing unauthorized pickup and eliminating the need for licensing.

The operational concept of the system is that of a hearing protector with provision for built-in background music and a paging override.

The functional concept of the system includes a control center-transmitter which selects any one of four widely available music sources (FM-AM tuner, phonograph, cassette, reel-to-reel tape deck), and frequency modulates the signal onto a carrier frequency. Depressing the push-to-talk switch on the paging microphone overrides the music source, permitting direct communication with anyone wearing a headset within the area enclosed by the loop. The output of the transmitter goes to a power amplifier, which provides enough current to drive two loops.

The use of frequency modulation (FM) is the key to the high quality of the David Clark Company's system. A person who is expected to wear a receiving headset for an entire eight-hour shift, should not be subjected to fading, static, distortion, or poor frequency response; although the system is monophonic, its response and distortion specifications rival those of stereophonic audio components.

The person wearing the headset is protected from the harmful effects of noise and also from the psychological irritation which occurs when listening to music for long periods of time, over poorly designed audio systems. The combination of an exceptional hearing protector, integrated with a high quality headphone-receiver, designed for long-term wear, requires the inclusion of a better grade music source in the system, in order to realize its full potential.

System Features

Wireless; no license required; high quality communications and music capabilities (monophonic); hearing protection; noise-attenuating headset with rechargeable Ni-Cad battery; recharging facility provided; accepts any standard music program source.

DESCRIPTION

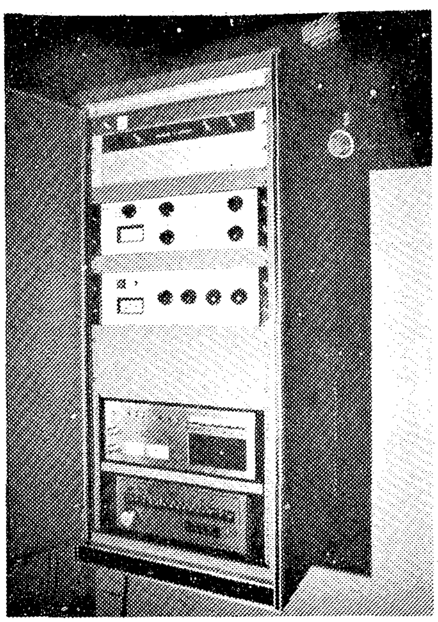


Figure 1

The David Clark Company Incorporated Wireless Headset System is designed to cover areas from 200 to 40,000 square feet, using a maximum of two loops.

The following are supplied with the system:

1. Cabinet - Designed to house Items 2 through 5 below and to provide security and proper ventilation. See Figure 1.

2. Transmitter

   a. Allows selection of up to four conventional music sources (phonograph, FM or FM-AM tuner, reel-to-reel, eight-track or cassette tape decks), and has separate level controls for microphone and music source.

   b. Separate level control for operator's monitor loudspeaker or headphones. (Loudspeaker supplied, headphones optional.)

   c. Large, illuminated VU meter allows setting optimum signal level to prevent distortion or noise.

   d. Once proper levels are set, paging microphone may be located remotely. Push-to-talk switch on microphone base activates microphone and suppresses music program for clearest communications. Music returns automatically when switch is released. (Microphone included; extension cords available.)

   e. Transmitter frequency-modulates all inputs onto carrier frequency for highest quality and lowest noise communications.

continued next page

THE DAVID CLARK COMPANY WIRELESS HEADSET SYSTEM, continued

3. Loop Tuning Module - Provides a means of interfacing loop (transmitting "antenna") with transmitter for maximum efficiency and coverage. Meter monitors transmitter carrier output for daily turn-on operational check and is used for setting tuning switches during simple one-time loop tuning procedure. Has provision for two loops, with potential coverage of up to 40,000 square feet; depending on construction.

4. RF Power Amplifier

   This item serves to boost the low level carrier output from the transmitter to a level high enough to driver the loops.

5. Music Source

   Up to four music sources of the customer's choice will be provided. (Any standard "stereo" or "hi-fi" component can be connected to the transmitter.) Stereo signals are converted to monophonic signals before being transmitted.

   An FM tuner or reel-to-reel tape deck can provide hours of unattended background music. An automatic turntable, eight-track and cassette decks will need more frequent attention.

6. Headset-Receiver

   a. Noise-attenuating, rechargeable battery powered headset provides hearing protection, paging, and background music.

   b. Built-in receiving antenna, on-off switch, and volume control.

   c. Operates up to twelve hours on overnight recharge.

   d. Plug accepts cable for recharging. See Figure 2.

Figure 2

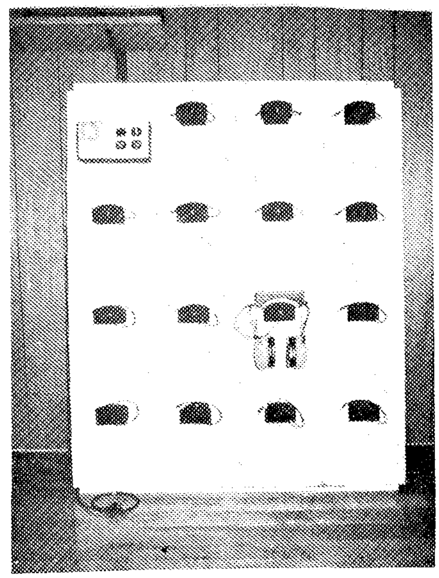


Figure 3

7. Charging/Storage Panel

   Provision for storing and recharging fifteen headsets.

   Has power supply and recharging cables built-in

   Interval timer allows setting of overnight (14 hours) recharging period with automatic shut off prevent overcharging. See Figure # 3.

AC -- DON'T TAKE IT FOR GRANTED

JOHN FREITAG of the AudioLab in Carbondale, Colorado and 2-time graduate of the Los Angeles class, sent in the following

We were installing an amplifier rack as part of a system at a local night club and the electrician had installed the 2 four-plex outlets exactly where we had specified. Nothing registered when I plugged in the electric drill and almost ripped my arm off with the resulting torque, nor did I get the message later when I plugged in the Porta-Vac to clean up the rack and nearly had a couple of manuals sucked up the tube along with the dirt. A bell finally rang as my assistant was soldering a couple of final ground connections prior to the initial testing. I looked over and noticed that the soldering pencil was a bright orange color. "Hold it," I said, "Get the Simpson". We put the meter on the outlet and 4 out of 8 were 220. It seems that the electrician had mixed up the neutral and hot on one of the boxes.

We were lucky that we had not plugged the amps in. It goes to show once again, you can't check part of something and assume that the rest is OK. Check them all.

MORE ON "FLOW OF CURRENT FROM POSITIVE TO NEGATIVE"

Karle S. Packard's answer to Paul W. Klipsch's question in the Letter to the Editor regarding which way electrical current flows is a classic in proving that "you should go to the source article first", something we failed to do in our answer in the October 1977 Newsletter (Vol 5, No. 1, page 21).

Legend?

Reginald Fessenden of Purdue University wrote a chapter on wireless in some encyclopedia that I read in 1919, but the name of which I have long since forgotten He explained the function of the vacuum tube, including heterodyne reception, and gave an elementary explanation of current flow within the tube where electrons flow from the hot cathode to the cold plate

I then asked my physics teacher to reconcile this with the teaching in our physics book (Millikan and Gale) that the "current" went the other way He admitted he didn't know, but said he'd "think about it." A couple of weeks later, he called me into his office (which gave me a scare), and reminded me of my question He said he still didn't know the answer, but that what might have happened was that Ben Franklin, when he flew his kite, assumed that the cloud was positive and the earth negative, and that the "current" flowed down the wet string He had two choices, and guessed wrong

The "flow of current from positive to negative" persists to this day The right-hand rule, symbolized by the IEEE logo, is accepted, and the electron flow in a vacuum tube is still in the "wrong" direction, to continue to confound the young physics student Does anyone have a better explanation? Is the Ben Franklin legend the way the dilemma really got started?

Paul W Klipsch
Klipsch and Associates Inc
Hope, Ark

Current flow

For a very good answer to Paul W. Klipsch's question concerning Ben Franklin and the direction of current in an electric circuit (June, p. 7) see "Electrolytic Conduction," Section 3-4 of *Introduction to Electric Circuits* by Herbert W Jackson

Mr. Franklin thought electric current to be a fluid*; and it is possible that, upon observing or reading a description of silver-plating, he deduced that the fluid (current) must be going from the silver anode (connected to the positive terminal of the battery) to the steel cathode (connected to the negative terminal of the battery), since the anode was losing weight and the cathode was gaining weight—and thus, the current carrying the metallic particles must be from the positive to the negative poles of the battery Mr Franklin was not aware of the fact that there is more than one type of electric-charge carrier

Leslie E Worden
Sinclair Community College
Dayton, Ohio

* Franklin's kite, *The World Book Encyclopedia*, vol 5, 1962, p 152

The horse's mouth

Rather than speculating on the origin of Franklin's choice for positive to negative current flow (Forum: P. W Klipsch, June, p. 7; L. E Worden, Sept., p. 24), reference to the relevant literature leads rather directly to the explanation Franklin first proposed the terms *positive, negative, plus,* and *minus* for describing degrees of electrification in a letter to Peter Collinson of London in May or June 1747. His terminology was obvious and straightforward: a body containing more of the "electrical fire" was said to be charged positively or plus, relative to one that had less. That more electrical fire did not turn out to be the same as more electrons was purely fortuitous, and was a consequence of the means for generating an electric charge.

The widely used generator of the time was a glass cylinder or globe that was rubbed, and in some cases was mounted on an axle for mechanical rotation This produced what was called vitreous electrification It was known that the opposite charge could be produced with other materials such as sulfur, in which case it was called resinous electrification The glass cylinder was, however, the most common by far, and it was this that Franklin used in developing his theory of electricity, thereby giving to its charged state the designation *positive*.

Regardless of the type of electrification chosen as positive, however, Franklin made the very natural assumption for direction of flow of electrical fire as being from *more* to *less*, or *positive* to *negative*. Thus if one reads Franklin's own words regarding his suggested terminology, it is perfectly clear why he chose the terms he did and they appear eminently sensible within the framework of contemporary knowledge.

For those interested in further investigation into this subject, I would recommend:

1. Cohen, I. B., *Franklin and Newton* American Philosophical Society, 1956
2. Roller D., and Roller, D. H. D., *The Development of the Concept of Electric Charge* Cambridge, Mass.; Harvard, 1954.

Karle S. Packard
AIL
Division of Cutler-Hammer
Deer Park, N.Y.

We had the above all put together for the January 1978 Newsletter but didn't have room for it. Soon thereafter WILLIAM McCAULEY of Sound Investments in Wichita, KS (Los Angeles class 1977) sent us the following: "I was interested in the letter from Paul Klipsch to the IEEE Forum, which you reproduced in the last Newsletter (October 1977). This letter sparked two replies from IEEE members which were reproduced in the same column. I have enclosed copies.

Also enclosed is a copy of some semiconductor daffy definitions, which I enjoyed, and thought you might also. I think I got it from the instrument division of the Birtcher Corp. Feel free to share it in the Newsletter.

SEMICONDUCTOR DEFINITIONS AND TERMINOLOGY

Holes--the presence of nothing
Hole Density--a concentrated amount of nothing in one small place
Electron--the absence of holes
Planck's Constant--about two board feet
Junction--fork in the road
p-n Junction--roadside rest area
Semi-conductor--truck driver
Degenerate Semiconductor--truck driver who likes his tea
Dope--someone you know
Heavily Doped--someone you wish you didn't know
Stored Charge--wine cellar
Silicon--gay prisoner
Germanium--would have been a flower, but someone misspelled it
Transport Factor--cousin of Max Factor
Majority Carrier--Republican with signs at a Republican Convention
Minority Carrier--Democrat carrying a sign at a Republican Convention
Base--low man in quartet
Common Base--low man in many quartets
Collector--one who collects
Common Collector--one who collects from everybody
Voltage Drop--a candy, like a gum drop
Atom--part of American colloquial expression "up and atom"
Delay time--the time one takes to start working after one has arrived on the premises
Rise Time--the time one takes to get up in the morning after the alarm has gone off.
Switching Transistors--the act of changing one transistor for another
DCTL--don't complain if the transistor's lousy

Business Week

Readers report

Metric coexistence

In "Canada goes metric" (In business this week, Sept. 19), you say that the U. S. is the only major trading nation that is not metric. Really now!

The U. S. is metric. It has been for 110 years by an act of Congress after the Civil War.

However, if your point be that all the world except the U. S. uses the metric system, then it must be pointed out in all fairness that 100% of the world uses the English system. The hard fact is that the two systems coexist everywhere, even in the most arrogantly "compulsory metric" countries.

B. C. Wiggin
Needham Heights, Mass.

A TIP (FROM AN ARCHITECT) TO SOUND CONTRACTORS DOING DISCO WORK

JERRY LAISERIN, president of E^{x}ponential Systems Corporation, Cranbury, New Jersey and graduate of the 1977 Philly class, makes a valuable suggestion to the disco contractor.

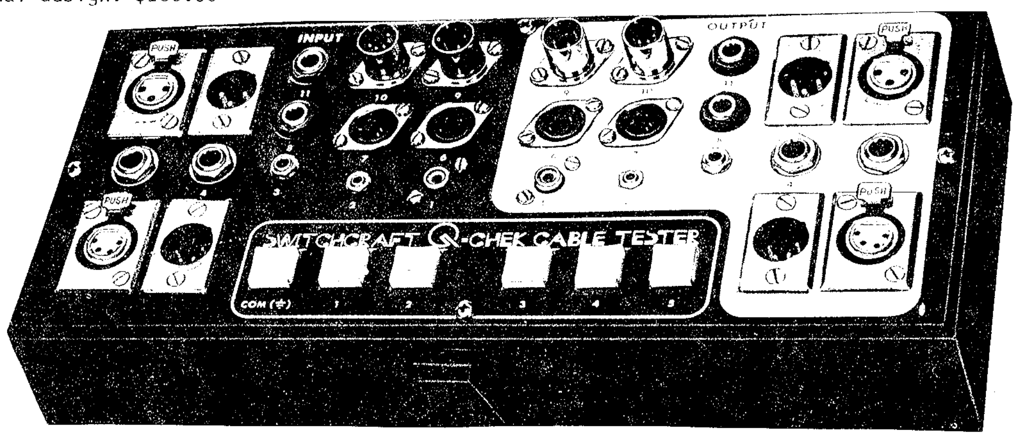
If you are fortunate enough to be involved in the project in time to review the architect's plans before construction don't overlook the details of raised dance floors, tiered seating platforms, etc., as these can be perfect bass traps unless the combination of joist depth/spacing, cross-bridging spacing, and subfloor thickness has been carefully studied.

Also beware of the "duckboards" for drainage of the floor behind the bar, especially if the bar is a long one. If the slats are too closely spaced in relation to their width and thickness, the result will be a series of slot resonators, punching another hole in the low frequency response.

The cost in additional lumber and labor to avoid these problems in the construction stage will probably cost your (and the architect's) client less money than the alternative: providing the additional amplifier power and speaker capacity at the affected frequencies to correct one or more 10-20 dB notches in a 120+ dB-SPL low frequency response curve.

OVERPRICED CABLE TESTER

A candidate for the most overpriced under-engineered test instrument is the Switchcraft Q-Chek. It can't even test the standard test cables used throughout the industry - the GR connectors or BNC connectors. It won't test patch cords of the original design. $186.00



CALCULATING THE THERMAL NOISE LEVEL IN DEGREES KELVIN

Thermal noise levels in audio are usually expressed as

-198 dB re: 1 volt, re: 1 ohm, re: 1 Hz or -174 dBm re: 1 Hz

A more universal expression for these values can be utilized by converting them into degrees Kelvin (°K)

In order to do this conversion you must first find the total noise power in watts (W_n) over the desired bandwidth in Hz (B_W) by measuring the noise voltage (E_N)

$$W_N = .001 \times 10^{\frac{\left[\left(20 \log\left(\frac{E_N}{.775}\right) - 6\right) - 10 \log B_W\right]}{10}}$$

Having found (W_N) the thermal noise limit is then calculated by

$$\text{TNL in °K} = 7.25 \times 10^{22}(W_N)$$

<u>Example</u>

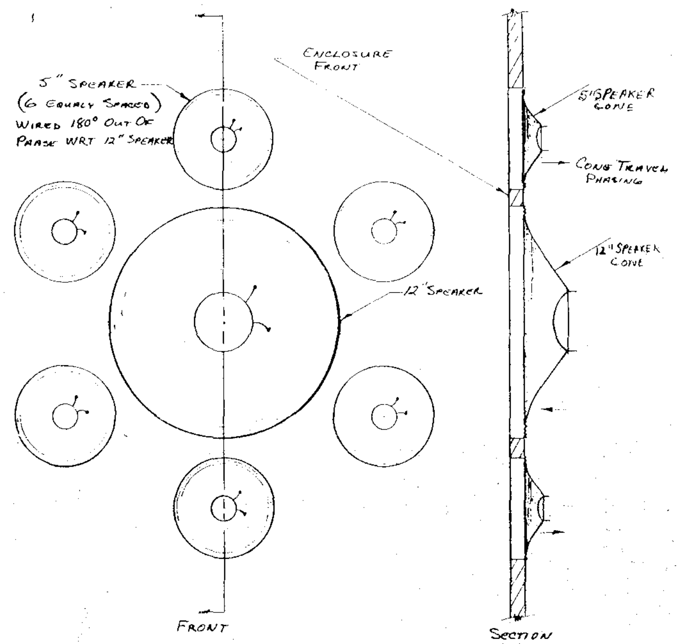
If you measured 438 uv over a bandpass of 20 - 20,000 Hz, then

$$W_N = .001 \times 10^{\frac{\left[\left(20 \log\left(\frac{438 \times 10^{-9}}{.775}\right) - 6\right) - 10 \log 19980\right]}{10}} = 4.02 \times 10^{-21} W$$

$$\text{TNL in °K} = 7.25 \times 10^{22}\ (4.02 \times 10^{-21}) = 291.13°$$

CAN YOU IDENTIFY THIS LOUDSPEAKER DESIGN?

KARL KROPP, Minneapolis class 1977, has located an interesting variation of loudspeaker driver arrangement. He asks, "Can any reader provide information on where this design came from, whose it is, etc.?" We'd all be interested to hear from you and to find a written article on the design of this unit.



RADAR SPEED TRAPS AND THE DOPPLER EFFECT

BILL SYMMES (San Francisco class 1973), now at Bogen, likes to drive fast BMWs and fly airplanes. We're sure the discussion from Bill which follows is intended only as an interesting technical discussion.

When listening to a sound from a moving source (or when you are moving) there is an apparent frequency shift either upwards (when coming towards you) or downwards. This is called the Doppler effect.

For example, a train whistle with a pitch of 250 Hz would sound like it had a pitch of 269.47 Hz if the train were coming towards you at 60 mph. If it were going away from you, the pitch would be 230.53 Hz.

$$f_1 = fo \left(\frac{C \pm V}{C}\right)$$

where f_1 = apparent frequency
fo = actual frequency
c = speed of sound
v = rate of closure (+) or separating (-)

The radar units employed by the authorities use the same doppler effect to calculate your speed. Radar travels at close to speed of light (186,273 mph) or (9.8352×10^8 ft/sec) but the same formula applies. For a car going 60 mph (88 ft/sec) a radar unit would see a return frequency of (assuming it transmitted 7000MHz)

$$f_1 = 7 \times 10^9 \left(\frac{88 + 9.8352 \times 10^8}{9.8352 \times 10^8}\right) = 7,000,000,623 \text{ Hz!}$$

A car going 61 mph would return a frequency of f_1 = 7,000,000,637!

So much for how they work - now what to do about it.

The traditional radar detectors currently on the market detect the presence of the frequencies used by the radar units. On a "straight shot" (ie. unobstructed) a good radar detector will give you about a 5-to-1 safety margin. This means if the radar trap is set to "clock" you at 500 feet, you should detect its presence at 2500 feet, giving you 2000 feet to slow down. Of course, it is not always a "straight shot" due to curves and hills.

Two newer types offer a greater challenge. First, the moving type can clock you from in front or behind, and from a moving vehicle - you'd really need two radar detectors for this. Second, there is a "trigger-operated" type which doesn't transmit anything until the trigger is pressed. Your radar detector will go off, but about all it's telling you is to reach for your wallet.

For some enterprising RF man, I offer a possible solution to this distressing situation. Aircraft have used for some time a device called a "transponder" which talks to radar systems. It is used to make the aircraft more visible on the radar screen and enables the radar controller to identify a specific aircraft.

When the transponder receives a radar transmission, it transmits an answer. This answer is so much stronger than the reflection that the reflection is not even seen on the radar screen.

Why not modify a transponder so it "answered" a radar unit with any desired frequency shift. The "answer" would block out any reflection and by being able to select the frequency shift, one could make the radar unit show any speed you wanted - regardless of the speed you are traveling!

PROBLEM SOLVERS FROM SHURE BROTHERS

A sound system engineer may or may not be a skilled circuit analyst. What a sound system engineer *must* thoroughly understand, however, in order to do his job efficiently is the basic circuit theory relevant to the interfacing of equipment, impedance matching or optimum adjustment, audio levels, polarity, and basic circuit configurations such as balanced, unbalanced, etc. He needs this knowledge in both choosing components suitable to system needs and in trouble shooting mis-wired or maladjusted systems. While the engineer's mind is the main tool, the experienced man accumulates many helpful devices with the passage of time.

Shure Brothers, Inc. have packaged a number of these tools, making them more convenient for inclusion in the typical tool kit. I carry the entire set and wouldn't want to do a sound system checkout without them along. (Note: most can't be used with simplex powered condensers.)

The three I use the most are:

1. Model A15LA Line Input Adapter. This unit allows me to plug 600 ohm unbalanced line level noise generators and oscillators into 150 ohm balanced microphone level inputs. Easy to see why it's *never* left behind.

2. Model A15PR Phase Reverser. The Phase Reverser allows quick polarity reversal by unplugging any cable and inserting this unit. It can save a lot of re-wiring when the reversal you are seeking does not lie in the immediate line under inspection but, say, in a mislabeled patch cord - which can happen.

3. A15TG Tone Generator. A really handy way to excite a a line during trouble shooting. Skilled ears listening to an oscillator can spot wrong levels, matches, etc., rapidly.

All the units listed can diagnostically save you a return to the shop by providing a quick comparison between two alternatives.

When we look at the professional's box we find Problem Solvers there.

Problem:	**Solution:**	
Input Overload	A15A	Microphone Attenuator prevents input overload Ideal where very strong signals are applied to a microphone input
Phasing	A15PR	Phase Reverser reverses the phase of a balanced line without modification of equipment
Low-Frequency Noise	A15HP	High Pass Filter provides a low-frequency microphone cutoff to reduce unwanted low-frequency noises and proximity effect
High-Frequency Noise	A15LP	Low Pass Filter provides high-frequency cutoff to reduce objectionable high-frequency noises.
Lack of Presence	A15PA	Presence Adapter adds voice-range intelligibility and extra brilliance.
Sibilance	A15RS	Response Shaper provides excellent sibilance filtering; flattens microphone response.
Line Level to Mic Input	A15LA	Line Input Adapter converts balanced low-impedance microphone input to line level input
Matching/ Bridging/Isolating	A15BT	Bridging Transformer, a balanced unit matches balanced or unbalanced devices of different impedances.
Troubleshooting	A15TG	Tone Generator produces a continuous 700 Hz low-impedance microphone level signal — extremely useful in setting-up and troubleshooting lines. Helps check levels, connections, mixer inputs, and cables. Allows one man to do the work of two!
Microphone Impedance Matching	A95 and A97	Series Line Transformers make it possible to connect low-impedance lines to mid- and high-impedance inputs (or vice-versa). Completely reversible. Solves problems of excessive high-frequency loss and objectionable hum

Size: 114mm (4½ in.) long x 19mm (¾ in.) diameter

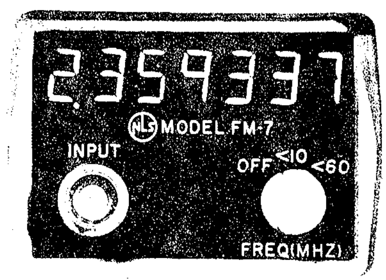
NLS FREQUENCY COUNTER

DOUG BROWN (1974 graduate) now with Non-Linear Systems in Solana Beach, CA recently delivered to us one of their new FM-7 60 MHz frequency counters.

We have now used it in three classes plus a great deal of laboratory work at home and have rarely, if ever, been more satisfied with an electronic device. This, in conjunction with my new battery operated H.P. 3466A digital multimeter, constitutes a fantastically accurate measurement of frequency and level in audio work.

The FM-7 has 1 Hz resolution on the low range (up to 10,000,000 Hz) which normally suffices for audio. This unit reads feedback frequencies with ease, doesn't become easily confused when several frequencies are present at one time but just fastens on the highest amplitude one.

PORTABLE BATTERY OPERATED



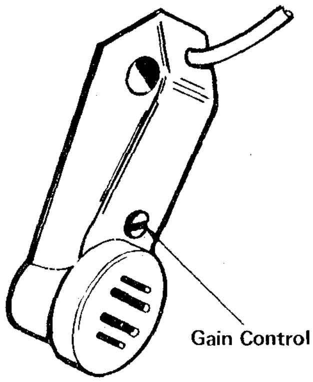
Only $195
With Rechargeable NiCad Batteries & Charger Unit.

The H.P. 3466A that I use in conjunction with the FM-7 frequency counter has features such as true RMS, AC+DC down to 10 uv ohms to 1 milliohm (with just two leads) and 1 uv DC, to name just a few.

Non-Linear Systems, Inc. can be reached by writing Box N, Del Mar, CA 92014 or call (714) 755-1134.

DAVID CLARK AMPLIFIED DYNAMIC MICROPHONE, MODEL M-2/DC

Many graduates have marvelled over the David Clark intercom we use in our microphone testing sessions. One of the most interesting features of this intercom is our ability to talk to you with our heads in a noise field of 120 dBA. The microphone system that allows this is shown on the specification sheet below.



AMPLIFIED DYNAMIC MICROPHONE, MODEL M–2/DC

for use with David Clark Company intercoms

The Amplified Dynamic Microphone is a noise cancelling microphone, self powered from the carbon microphone exciting source, and designed as a direct replacement for carbon microphones. It is not polarity sensitive and gives the best noise cancelling performance when the microphone touches the user's lips. The screwdriver adjustable gain control is normally set at maximum but can be turned down if the intercom electronics overload. The Amplified Dynamic Microphone is compatible with David Clark Company wire booms.

SPECIFICATIONS -- MODEL M-2/DC

CORD AND PLUG:	MICROPHONE P/N 12987G-01, WITH 10" (250 MM) TWO CONDUCTOR WD34/U CORD AND U173/U PLUG MICROPHONE P/N 12987G-02, WITH 17" (430MM) TWO CONDUCTOR WD34/U CORD, WITHOUT PLUG
EXCITATION:	24 VOLTS BEHIND 1800 OHMS
FREQUENCY RESPONSE:	GAIN MINIMUM: ±2dB 250-4000 HZ GAIN MAXIMUM: RISING AT 3 dB/OCTAVE TO 4000 HZ. THEN FALLING AT 15 dB/OCTAVE
GAIN CONTROL:	9 dB VARIATION OVER 270° ROTATION. CLOCKWISE IS MAXIMUM GAIN.
OPERATING DISTANCE:	ZERO TO 1/4 INCH FROM THE LIPS.
NOISE LEVELS:	OPERATES WITH A MAXIMUM AMBIENT NOISE LEVEL OF 120 dB.
SENSITIVITY:	40 MV INTO 1800 OHM LOAD, 24 VOLT SUPPLY, 1000 HZ AT 100 dB.
DIMENSIONS:	62X19X19 MM
WEIGHT:	40 GRAMS

THE EQUIVALENT LEVEL (L_{EQ}) IN NOISE MEASUREMENTS

Increasingly,acoustical workers in the noise control field, funded by federal, state, and in many cases, even smaller political subdivisions, are erecting an interesting edifice of measurement systems.

A number of these measurement systems are based on the concept of average energy. Suppose, for example, that we have some means of collecting all of the A-weighted sound energy that arrives at a particular location over a certain period of time such as 90 dBA for 3.6 secs (this could be a series of levels that lasted seconds, hours, or even days). We can then calculate the decibel level of steady noise, for say, one hour that would be the equivalent level of the 90 dBA for 3.6 secs. That is, we wish to find the energy equivalent level for 1 hour

$$L_{EQ} = 10 \log_{10}\left[\int_{f_1}^{f_2} PA^2 \, dt/(Po^2(T_2 - T_1))\right]$$

In our example this integration reduces to

$$L_{EQ} = 10 \log_{10}\left[\frac{10^{\frac{90}{10}} \times 3.6 \text{ secs}}{3600 \text{ secs}}\right]$$

Thus, one hour of noise energy at 60 dBA is the equivalent energy exposure of 90 dBA for 3.6 secs.

L_{DN} (day-night level, CNEL (community noise level), etc., all follow similar schemes with variation in weightings for differing times of day, etc. Syn-Aud-Con Tech Topics Vol 3 # 6 and Vol # 3, No. 10 provide further detail on these measurements and calculation techniques.

It's of interest that shooting a .458 magnum 174.7 dB-SPL (peak) for 2.5 milliseconds translates into

$$L_{EQ} = 10 \log_{10}\left[\frac{10^{\frac{174.7}{10}} \times .0025}{3600}\right] = 113.12 \text{ dB}$$

of steady sound for 1 hour. OSHA allows only 15 minutes of exposure to levels of 110 to 115 dBA (one of the few times I have agreed with OSHA). As Howard Raurk's African guide, Harry Selby, remarked after Ruark had accidently set off both barrels at once of a 470 express rifle while being charged by a Cape buffalo, "One of you ought to get up."

GRIN

Just because you're a paranoid doesn't mean that there isn't someone out to get you.

Trespassers will be violated.

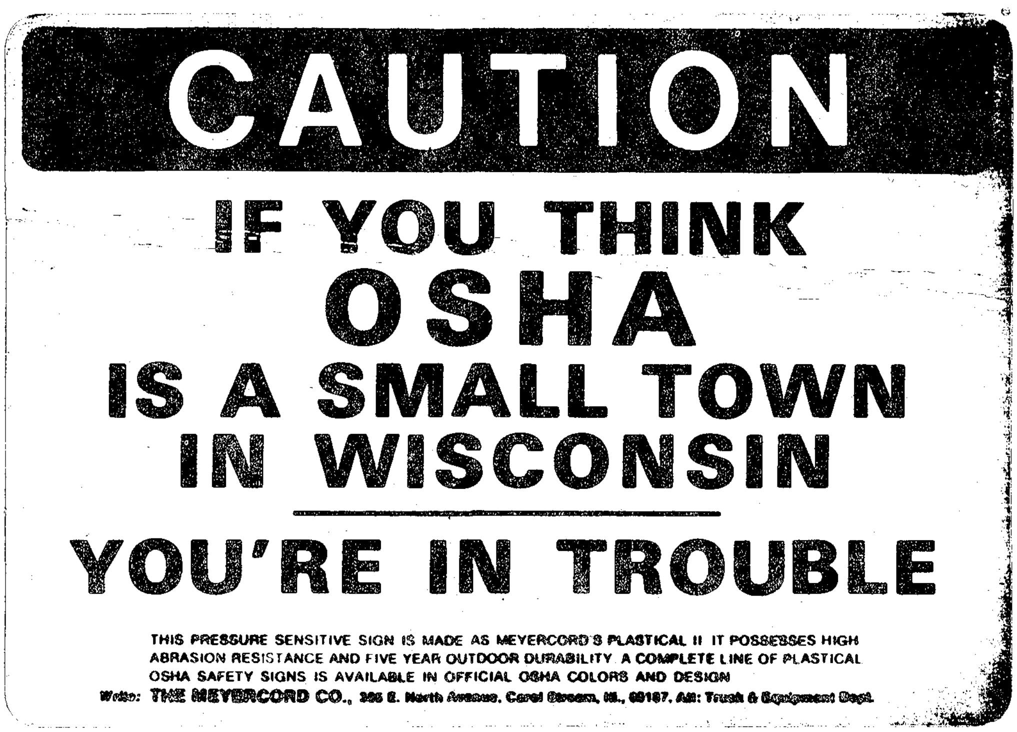
Anyone found here after dark will be found here in the morning.

Beware of dog. Trespassers will be eaten.

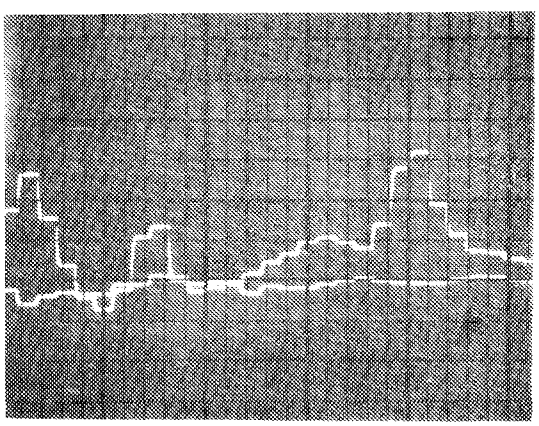
JERRY LAISARIN, AIA, in our 1977 Philly class shared a beautiful story, worthy of Don Petros: On the Ark, Noah tells the pairs of animals to multiply. Two snakes say they can't multiply because they are adders. Noah deals with the problem by putting the snakes on a rough sawn wood table. Now the snakes can multiply because as we all know, Adders can multiply if they use a log table. (from Dr. Dobb's Journal of Computer Calisthenics & Orthodontia)

From STUDIO magazine: According to a recent item in the "America" columns of the London Daily Mail, the latest gimmick in the stateside burial market is talking gravestones. When mourners get near enough to the grave of their loved ones it automatically spouts a synthesized message like "I'm Jane Smith. I died on June 16, 1976. Thanks for coming to see me". The company that makes these little goodies says there is such a big order, it can't build them fast enough.

KEITH MORTON, P.E., two-time graduate from Richard-Gebaur AFB, sent this sign:



AN INTERESTING IMPEDANCE CURVE

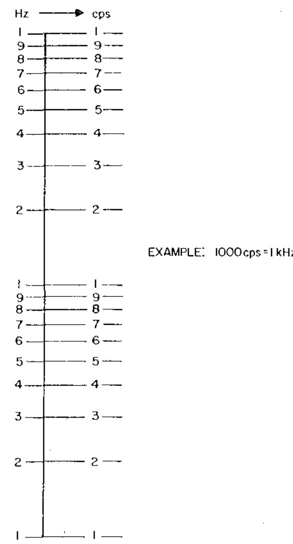


BURT BOETTCHER of Ken-Com Engineering says he just couldn't resist "doctoring" a cartoon he saw in an electrical magazine.

Nomograph accurately converts Hz into cps

Sir:

The accompanying nomograph was designed to facilitate conversion from hertz into cycles per sec-



ond. Although as shown it spans only two decades, it can be extended to higher and lower frequencies by making use of the well-known relationships, $10^{-x}\ 10^{x} = 1$ and slipping the decimal point.

Another nomograph is now in the process of validation by extensive computer runs. It will mechanize the conversion from cycles per second into hertz.

Guy Fawkes
Fleagle Design, Inc
Boston

BOOK REVIEWS

SHOCK AND VIBRATION HANDBOOK (second edition) Edited by Cyril M. Harris and Charles E. Crede, published by McGraw Hill 1976. $32.50

This massive 1322 page, 1025 illustration, single volume with 44 chapters written by 54 different authorities actually is intended to replace the original three volume edition for those not desirous of collecting the archival material the first edition contained.

One chapter of immediate interest to me was the last chapter, Chapter 44, *Effects of Shock and Vibration on Man"* and it should be of interest to anyone designing sound systems for Discos. It is written by Henning E. Von Gierke and David E. Goldman. Included in Chapter 44 is a five degree of freedom body model, Fig. 44.14 (useful in deciding how to shake what) and a Table of Criteria for short exposure to vertical vibration for various frequencies which, among seven symptoms experienced, included the number of subjects out of a test sample that experienced testicular pain at frequencies from 1 Hz to 15 Hz (10 Hz definitely did the job for the majority of males).

Maximum impacts, Gs, etc., all are discussed and illustrated in detail. Specifics about Col. Stapp's 40G deaccelerations are presented. You can, for example, if properly belted in (crotch, lap, chest, and shoulder harness, such as used in contemporary race cars) survive without injury 40 to 50G for up to 100 msec, and survive with some damage up to 100 Gs. In contrast, a seat belt of the type standard in a passenger car lets you survive (but with injury) only 10-20 Gs.

The displacement of the abdominal Viscera occurs between 3 and 3.5 Hz, which means that the police loudspeaker discussed in class will really have to be a big one. Four to 6 Hz and 10 to 14 Hz are the resonant frequencies of the pelvis and lower spine.

The chapters are divided broadly into three categories: (1) Theory and fundamentals, (2) Instrumentation and measurement, (3) Equipment design, packaging, and practical solutions to typical problem areas.

The text is not always easy reading but the illustrations are exceptionally useful. For a book this size and with as many distinguished contributors and able editing, $32.40 is a bargain price - and a worthwhile basic text in a serious engineer's personal library.

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HIGH PERFORMANCE LOUDSPEAKERS by Martin Colloms, published by Halsted Press, a division of John Wiley & Sons Inc. $22.50.

It seems to me to be a grossly overpriced book. Contains only 237 pages, it is essentially a well written book but repeatedly reflects the author's electronic circuit training and lack of acoustic training. He, for example, thinks loudspeaker sensitivity and efficiency are just different words for the same thing. He has apparently never heard of Q, and mistakes physical alignment of drivers for Time Align™.

Still, he does provide an interesting survey of where the British hi-fi market is headed in terms of loudspeaker systems and does provide insight into how many English loudspeaker designers are employing the work of Thiele and Small.

Overall judgment - a component book with patches of useful detail. Because of a total failure to appreciate the loudspeaker system problem it is of extremely limited value to an audio systems engineer.

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PHYSICS OF STEREO/QUAD SOUND by Joseph G. Traylor, published by Iowa State University Press, Box 235B, Ames, Iowa, 50010. $9.50.

Dr. Traylor combines enthusiasm for music with respect for physics in a readable and informative manner. I particularly liked his approach to Chapter I, *Fundamentals of Sound Basic to Physical Laws*. At $9.50 it is well worth recommending to young people desiring insight into audio engineering.

"THE NEW NEW TELEPHONE INDUSTRY"

Mother Bell has 120 billion dollars of installed phones, wires, and switching equipment. More than 70 million homes in the US are wired into the telephone network. Last year the US telephone industry billed its business customers more than 25 billion for telecommunications services. At the same time the US data processing industry generated 38 billion in sales.

What's fascinating to all observers of these two giant industries is the area where they overlap. Communications-based information systems is the fastest-growing market segment for both telephone and computer companies and one that could easily double and re-double in the next decade. Leaders in the industry forsee the market as a 300 billion dollar a year business in the near future.

IBM is selling a large electronic PBX in Europe and some see that as a rehearsal for the main show in the US. IBM's cash hoard of 5 billion dollars, and its size -- annual revenues of 18 billion dollars -- make it the most formidable competitor that AT&T has ever had to face.

Since neither Congress nor the FCC is content with the present chaotic mix of regulation and competition, and neither wishes to see a potential 300-400 billion dollar a year industry split between just two giants, regulated or not, big trouble could be just around the corner.

All of this and a great deal more is discussed in the article, "The New New Telephone Industry", in the February 13, 1978 issue of Business Week , pages 68-78.

Getting a handle on two giants this size reminds us of W.D.M. Bell's remarks about hunting man-eating lions, "I believe most hunters are injured by lions because they fail to coolly place their shot with great accuracy" and a few pages later he says, "There are easier things to hit than a charging lion." (These words have saved many of us an African trip.) The realization that our congressmen and government officials "are the best money can buy" might make the charging telecommunications industry very hard to hit. This system has worked well in the past, so I'm not criticizing it, merely philosophizing.

DBD

ARTICLES OF INTEREST

New York City graduate, GARY HARRIS of G&T Harris, Inc., is an exceptionally talented theater sound engineer with considerable experience in the reinforcement of legitimate theater, opera, and the myriad of modern traveling entertainers.

Gary has written an eminently readable article in the Jan-Feb. 1978 issue of Theatre Arts entitled, *KEEPING "GOOD BUDDY" OFF THE STAGE*. The article, as the title suggests, is on wireless microphones. One of the fascinating possibilities opened by the use of wireless microphones is the following quote from the article:

> Untunable transmitters and receivers create other problems. On a recent jaunt by an otherwise well prepared star (Liberace) the decision was made to take along only two of his normal four transmitter groups. As luck would have it there was a mix-up and no transmitter frequency matched any receiver frequency -- a situation that was impossible to correct when it was discovered on stage.

Wireless microphones are like hunting lions with a .22, it can be done but you have to like danger.

DBD

CLASSIFIED

FOR SALE:

Texas Instruments SR-52 magnetic card programmable calculator, the usual accessories, the basic program library (22 cards), and about two dozen programs in acoustics, noise control, sound system design, image projection, auditorium design, etc., $175; with PC-100A printer $350.
TED UZZLE, Box 718, Cambridge, MA 02139. Ring (617) 492-5585 during business hours EST

FOR SALE:

Test Gear: Amber 4400 Test Set, one year old $2300. Philips PM 3232 Dual Beam, 10MgHz Scope plus two 9327 Probe sets (1:1 and 10:1) plus 1 Polaroid CR 9 Scope Camera, $900. All for $3100.

RICHARD JENNINGS, 209 Arbor, San Francisco, CA 94131. (415) 586-9914.

FOR SALE:

16 Shure voice gates with 4 rack-mounts. Modified for use with NOM (number of open microphone) circuit. Make offer for part or all.
GORDON WOLFE, OKO Electronics, 2526 Loma Ave, South El Monte, CA 91733. (213) 579-5100

FOR SALE:

HP-97 with accessories and programs. Make offer.

JOHN ODUM, The Music Mart, Pennyrile Mall, Hopkinsville, KY 42240. (505)885-5386

WANTED:

Sound, installation/service technician.

ALEX ROSNER, Rosner Custom Sound, Inc. 11-38 31st Ave, Long Island City, New York 11108. (212) 726-6600

WANTED:

Lockwood Sound Symbol Templates. Any reasonable price will be considered.

BRUCE THAYER, WMT Stations, Box 2147, Cedar Rapids, Iowa 52406. (319) 395-6000, 8:00-5:00, Monday-Friday. Call collect.

LOOKING FOR EMPLOYMENT:

Seeking a position as a designer/installer of commercial audio or multi-media systems in the Chicago area.
ROBERT V. KRUGER (Chicago 1977 class) 407 W. Ethel Ave, Lombard, IL 60148

LOOKING FOR EMPLOYMENT:

Open to a wide variety of jobs in the audio field -- design, sales, construction, installation, engineering, etc. If the situation is right, willing to relocate.

STEVEN CHARLES MacNEAR, 1206 Covell Place, Davis, CA 95616. (916) 756-1321

LOOKING FOR EMPLOYMENT:

Seeking work as a sound system designer. Wants to work for a commercial sound contractor. Graduating this June in E.E. Worked part-time in audio. Will relocate.

EDWARD J. SPITZIG, 521 Pine Street, Aptos, CA 95003. (San Francisco Class 1978)

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