"SYN-AUD-CON"
SYNERGETIC AUDIO CONCEPTS

newsletter

P.O. BOX 1134, TUSTIN, CALIFORNIA 92680

VOLUME 6, NUMBER 1
October, 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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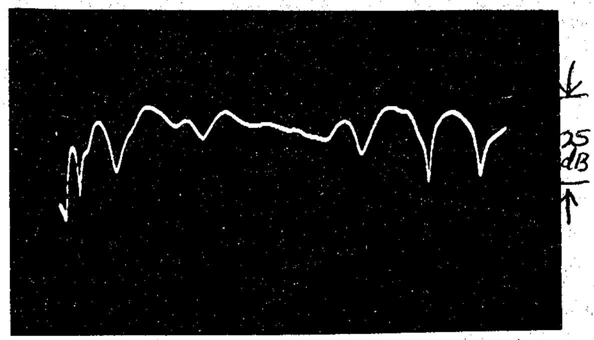
REPORT ON TDS, LEDE, PZM AND SYN-AUD-CON

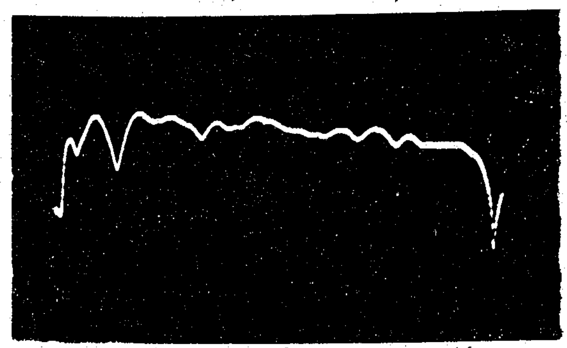
In the short time interval between the July Newsletter and this Newsletter, a series of events have occurred surrounding our work with TDS in recording studios and their control rooms.

Early in the year we began an investigation into how TDS might be made more accessible to those engineers in the recording industry alert to TDS's potential. One conclusion we had arrived at through working with CECIL CABLE during the 1977 classes was that TDS has only limited use in large auditoria, arenas, etc, in terms of their reflective properties but that the real potential was in uses such as

1. In situ acoustic absorption measurements
2. The study of loudspeaker misalignments. (Some manufacturers have real serious difficulties in this regard - far more serious than we at first realized. See photos of a commonly used studio monitor.)
3. Reflective properties of small dead rooms - particularly control rooms.
4. Calibrating PZM™ microphones
5. Measuring loudspeaker Q rapidly
6. Studying the presence or absence of reverberant sound fields.



HI. FREQ UNIT IN OPERATION



HI FREQ UNIT BLOCKED BY HAND

We now have in excess of 25 licensees to practice TDS. (We, being California Institute Research Foundation.)

Cable Bros. (Cecil Cable)
Marshall Buck, Ph.D.
SYN-AUD-CON (Don Davis)
Acoustilog (Al Feierstein)
Jamieson & Assoc.(Richard Jamieson)
WED Enterprises
Ken A. Wahrenbrock
MTS Northwest, Inc, (Ed Lethert)
Timothy J. Clark
William M. Peterson, Jr.
Capitol Records (Richard Blinn)
Westlake Audio (John Payne)
U.R.E.I. (D. F. Morris)
Martinsound (A. J. Martinson)
John J. Klanatsky
G & T Harris, Inc.
Eugene T. Patronis, Jr.
Joe Mitchell
Andrews Audio Consultants (David Andrews)
Nelson Rose
Glen Ballou
Heavy Custom Sound & Light (Richard C. Coscia)
JFA Electronics, Inc. (Robert E. Brown)
Rayburn Electronics (Ray A. Rayburn)
Dale Ashby & Father (John Laberdie)
Robert V. Vitale
Klipsch & Associates
Farrel M. Becker
Acoustic Control Corp. (Ed McGee)
Valley Audio (Bob Todrank)
Marguerite's Music (Don Mowry)
Masstronics, Inc, Robert Daniel
Quality Sound Enterprise, Mark N. Lynch

We are very proud of the people on this list.

They are using a new analyzer of much smaller physical size and 1/3 the price of the unit we demonstrated in class last year. This new analyzer, off-the-shelf and rentable-by-the-month, possesses greater detail and a digital memory and is much easier to use. Two steps are necessary in order to receive from us full details on how to go about using TDS.

The first step is to send to Syn-Aud-Con's Tustin office a check made out to the California Institute Research Foundation for $100. The second step is to send at the same time a check made out to Syn-Aud-Con for $5 that covers the special package of data we have prepared for the licensees. We will then see that you receive your license from Cal Tech and full data on what equipment to buy and how to use it.

The excitement is not in the fact that TDS is now much more accessible and easier to use but rather in *how* it's being used to revolutionize recording techniques.

LEDE CONTROL ROOMS

CHIPS DAVIS was the first to take the plunge and redo a control room so that the rear half (behind the mixer) is hard, live, and diffuse, and the front is nearly anechoic. The rear wall (hard as it is) is acoustically invisible - no sound comes off of it to the mixer's ears, thanks to the Haas Effect.

Since the front area can cause no early reflections, there are no broadband anomolies to radically "color" the sound and identify the control room. Thus the mixer hears at the console the same sound he hears out in the studio with his head placed between the Pressure Zone Microphones (PZM™). Naturally loudspeaker coloration still enters in but with the room coloration removed, it is no longer masked, but "out in front". Time Align™ monitors come into their own in the LEDE environment.

Needless to say, Chips has had a literal pilgrimage to his studio and there are several LEDE control rooms on the drawing board.

THE PRESSURE ZONE MICROPHONE

Those graduates attending the special graduate meeting in April witnessed the first rough attempt at PZM by Syn-Aud-Con. About three weeks later Syn-Aud-Con graduate Ken Wahrenbrock,who had attended the graduate meeting and had read "Putting it All Together in the Control Room" which contained a drawing of how we thought Ed Long and Ron Wickersham might have mounted their PRP microphones for their excellent Reference Recordings, walked into our Los Angeles class with a first-cut attempt at our drawing and said, try this.

A report, continued

In every demonstration that we have performed, the PZM system is preferred and this includes comparison with microphones costing up to $1,000 a piece. Ken's PZM system sells for $200 for a stereo pair. Ken's price includes the optimum power supply, transformers, cable, etc. Also, the price includes a licensing fee to Ed Long and Ron Wickersham.

Syn-Aud-Con has applied for a trade mark for Pressure Zone Microphone. We feel that PZM describes the physics of the system that Ed Long and Ron Wickersham call the Pressure Response Pickup (PRP™). A pressure response microphone may or may not be used in a pressure zone. A PZM system creates its own special pressure zone.

Recordings literally start from ground zero again with the advent of these microphones. Syn-Aud-Con feels that those of you who do recording and sound reinforcement work and fail to obtain one of Ken's systems are missing out on a great opportunity.

SUMMARY

It is important to remind our readers that Syn-Aud-Con has no financial involvement in TA™, LEDE, PZM™, or TDS. We repeat - while we have been instrumental in the sharing of ideas that led to these developments and in the case of LEDE literally developed the fundamentals, we are in no way financially benefited by your licensing to TDS or your use of these new discoveries.

We know that these discoveries are destined to change the recording industry and we want our graduates to lead in this change. PZM is also, we feel, the ideal choice for a majority of the sound reinforcement applications, and TA loudspeakers used with PZM systems eliminate many of the causes of violent feedback in such systems.

We feel that Syn-Aud-Con will, over the years, lead to many new discoveries in our industry. We're eager that you all share in the thrill of these current ones.

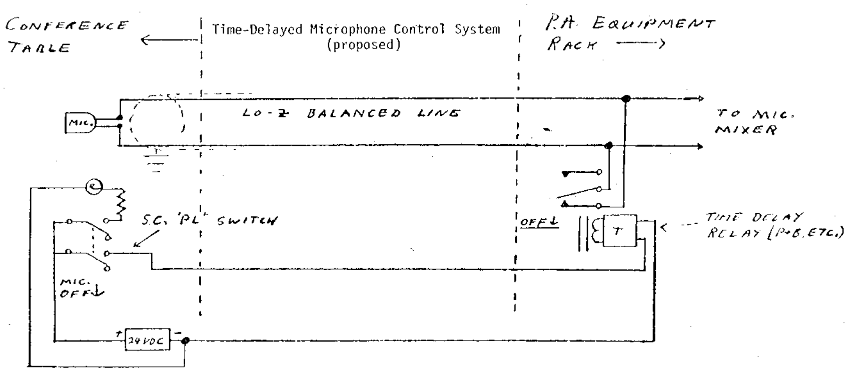
REDUCING MICROPHONE SWITCHING NOISE

CHARLES TOWNSEND, communications engineer for the state of Florida sent in the following material and invites your comments: (State of Florida, Department of General Services, Larson Building, Tallahassee 32304.)

In the Syn-Aud-Con Newsletter for January, 1978 (Vol 5., No. 2) a quiet microphone on-off switching technique as submitted by Bob Reim was presented on page 10 of that issue. This switching technique has been utilized in all our designs for conference room sound systems in the new State Capitol Complex with considerable success as it generates very little (if any) electrical noise when the switch is operated.

Problems of an acoustical nature can arise, however, if the system design dictates the use of lighted push button switches for each microphone.

All switches which we have evaluated to date produce an audible "click" when operated. The intensity of the click varies with switch design. Switches measured ranged from levels of 68 to 71 dB as indicated on our GR 1933 SLM (impulse mode) at a distance of 6' from the SLM mike. The Altec 8050 RTA indicated that most of the "click" noise was being generated between 3.16 and 8 KHz. For both tests the switches were hand held tightly against a table top (to simulate mounting in a conference table) the lower frequencies and overall perceived loudness of the switch operation increased markedly. When the switch is mounted next to a sound system microphone, the switch noise is transferred to the mike by 2 methods.



1. The "sounding board" formed by the table transfers the sound to the mike mount, into the mike case, to the mike element.
2. The click travels through the air some short (6-18") distance and is picked up again by the mike.

The problem, therefore, is acoustical in nature. We have devised several possible solutions, some of which are electrical and some of which mechanical. The mechanical approach (shock isolation) would require a sacrifice in mechanical rigidity and/or the physical appearance of the switch/mike combination, especially to the mikes that are mounted on gooseneck extensions (to increase D_S). In the system under consideration, our client would not accept compromises in this direction.

The proposed electrical solution, therefore, is to use a Switchcraft PL series switch and a time delay relay. The relay should be set for a delay of 0.5 to 1.0 second on pull-in to allow all switch noise to die out prior to the mike becoming "hot". Relay drop-out will be essentially instantaneous to re-short the mike line before the switch push button returns to its outward stop (generating an audible click). See drawing above.

Some of the switches used in existing systems are of a snap action design and upon "mike-off" activation of the switch will generate a click before the relay can drop out. The construction of the Switchcraft PL series switch, however, features a much slower and quieter movement of the switch mechanism which, in our opinion, will allow the relay to drop out in the required length of time.

THE PIONEER LEDE CONTROL ROOM

It takes the pioneer spirit to open thought to the truth and allow it to lead into unexplored areas. CHIPS DAVIS was in our February Anaheim class and heard our theoretical discussion of the gathering evidence that TDS was providing regarding frequency response anomolies caused by the mixing of early reflections with the direct sound from the monitors.

With only this preliminary theoretical discussion, Chips planned a Live end-Dead end (LEDE) control room for his Las Vegas recording studio. Some pioneers turn back, some get a rear end full of arrows, but Chips not only crossed the desert safely, he arrived at the Mother lode.

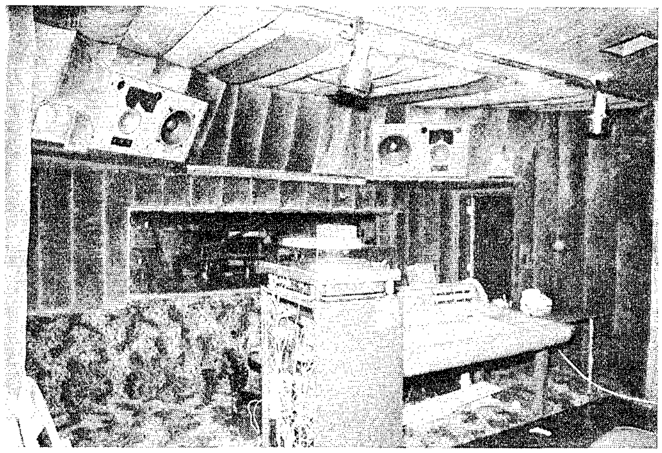
We received an excited phone call from Chips telling us "I've done it". "Done what", we asked? "Built your live end-dead end control room" replied Chips. Naturally we got on a plane for Las Vegas as soon as we could. We, being Don, Carolyn and Bill Putnam, owner of United Recording, UREI, etc.

What We Heard

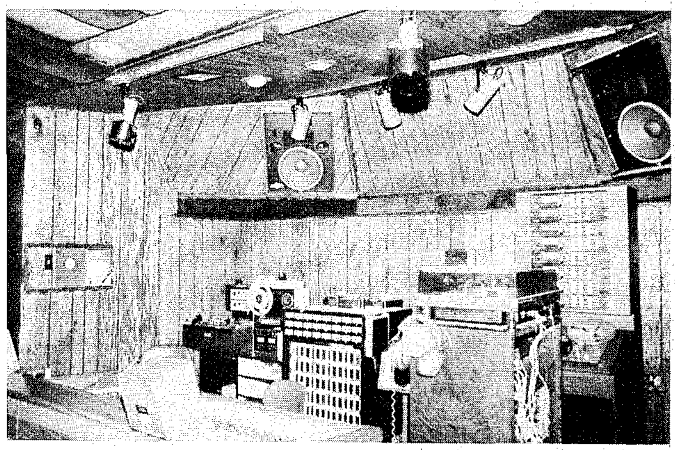
Surprise number one is that while the rear wall is hard hard hard and carefully angled for maximum diffusion, *IT IS INAUDIBLE* at the mixer's position. You can turn and face the hard rear wall, cup your ears and none of the sound from the monitor loudspeakers appears to come from the wall. NONE! Even with your back turned to the loudspeakers you clearly hear them behind you and NOTHING from the rear wall.

Why?

Remember the demonstrations of the Haas Effect in class? Twenty milliseconds is the magic number for creating the illusion that all the sound comes from only one of the two speakers. If we consider the sound from the monitor loudspeakers at the mixer's ears as one acoustic source and the rear wall of the control room as the other acoustic source (virtual source displaced in time) and the distance from the rear wall to the mixer is approximately 20 milliseconds (double path) or ten feet to the wall and ten feet back as measured from the mixing engineer's ears, we have created the Haas Effect. Thus the rear wall simply disappears psychoacoustically, in terms of detectible direction, while continuing to provide a highly useful increase in overall sound level.

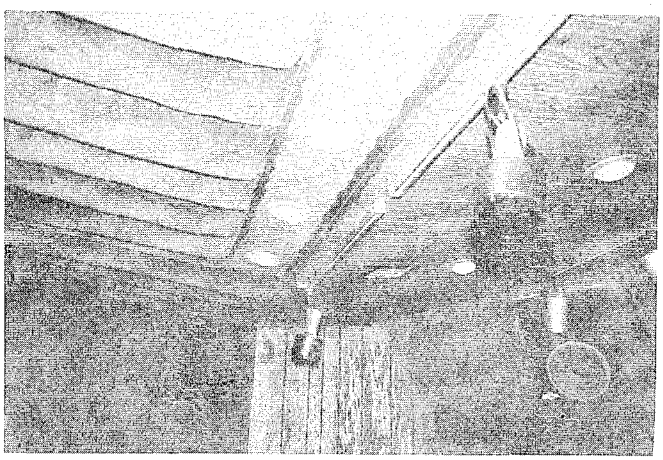


Front of the control room



Back of the control room

Turning towards the front wall, which is made as non-reflective as possible, we heard only the monitor loudspeakers minus the usual coloration effects. Subsequent investigation has shown this is largely due to reflections with path differences longer than the direct sound of 1 foot or less. It was easy to establish that what we were hearing at the mixer's position was the same as what was heard in the studio when we placed our head between the microphones (we were using the PZM system.) What has been achieved is a really major step forward in the reduction of the influence of the control room on the sound heard from the monitor loudspeakers.



The absorptive front of the control room ends directly behind the console

Close-up of the front of the control room

Chips Davis, continued

What Kind of Anomolies are Present?

Very *narrow band* anomolies are the kind generated when the path difference between the direct sound and the reflected sound exceeds 10 feet

$$B.W. = \left(\frac{V}{D}\right)12$$

Where D is in inches
V is the velocity of sound (1130 ft/sec)
B.W. is bandwidth in Hz

$\left(\frac{1130}{2x120"*}\right)12$ = 56 Hz or approximately 1/6-octave at 500 Hz; 1/12-octave at 1,000 Hz, etc.

*10'=120"; double path = 2x10'

It becomes very obvious that care must be exercised to insure that no combinations of rear wall reflections possessing slightly differing path lengths be allowed to combine at equal levels at the mixer's head position.

Summary

These photographs reveal what the first LEDE control room looks like. At this point in time it is considered a substantial improvement in control room acoustics by those who have had an opportunity to listen to recorded material being played back in the control room. What is not yet known is "Are there ways to substantially improve the advances already made?" And the answer has to be "of course!"

J. W. DAVIS COMPANY

A small company we encounter quite frequently when talking to large distributors is the J. W. Davis Co. of Dallas. Their president, HARVEY EARP, was in our last Dallas class and in the special April 30 graduate meeting, so we have become better acquainted with the company, which is now in its "44th year of service".

They have a catalog that contains some very hard-to-find items such as really remarkable inexpensive wall baffles made out of structural ABS material; an inexpensive but perfectly effective amplifier coupler; and a useful 10-watt tube amplifier. For the beginner in sound, they supply everything needed except the hand tools. For the multi-line professional they can provide those hard-to-find special items in baffles, wall plates, small pads, etc.

The J.W. Davis Company is not related to our branch of the family but they have said they will send copies of their excellent 65-page catalog (normally costs $2.00) to any Syn-Aud-Con graduate, at no charge, who requests it. The J.W. Davis catalog is definitely worth having on your reference shelf.

BELL TELEPHONE'S ORIGINAL DEFINITION OF THE T.U.

In 1924 the Bell Telephone Laboratories converted from the "standard cable mile" (used for twenty years as a measure of gain and loss) to the Transmission Unit (T.U.) The T.U. was defined as

$$T.U. = \frac{\mathrm{Log}\ \frac{P_1}{P_2}}{\mathrm{Log}\ 10^{0.1}}$$

Any two power ratios were compared logarithmically with a reference level of $10^{0.1}$ or $10^{1/10}$ power.

In 1929 this same unit was renamed the decibel (dB) and was then written as

$$T.U. = \frac{\mathrm{Log}\ \frac{P_1}{P_2}}{\mathrm{Log}\ ^{0.1}} = \frac{\mathrm{Log}\ \frac{P_1}{P_2}}{1/10} = 10\ \mathrm{Log}\ \frac{P_1}{P_2} = dB$$

In writing about the T.U., W. H. Martin made the following, to us, still pertinent statement, "In considering the conversions between sound and electrical energy, it is *obviously advantageous* to have a unit based directly on a *power* ratio." Journal of the IEEE, June 1924

Chinn et al in 1939 further wrote, "If the fundamental concept were voltage, apparent gains or losses would appear whenever impedance - transformation devices, such as transformers, occur in a circuit. This difficulty is avoided by adopting the power concept." (A 1939 paper describing the new standard for volume indicating meters worked out by NBS, CBS, and BTL)

MEL SPRINKLE has written us to ask that we *stress* that the dBm is merely another way to say *WATTS*. +30dBm is 1 watt. 1 watt is +30 dBm. Mel feels this simple fact is often not sufficiently stressed and we tend to agree after reading the recent rash of dBm_e proposals, etc., that are being presented to the young and innocent.

QUESTION: DO YOU KNOW YOUR dB?

I measure 2 volts at the input of a black box and 1 volt at its output across a load. Does the box have gain or loss? (Answer elsewhere in Newsletter)

PRESSURE ZONE MICROPHONE (PZM™)

Among conventional microphones (and the world is now divided into BCM - before coherent microphony) and ADCM (after discovery of coherent microphony), we have always rated the Shure SM-7 as one of the very best. It is uniform in free field response, has exceptionally good polar response, and has very low distortion. As we know, even a perfect free field microphone exhibits gross response anomolies when placed near a reflective surface.

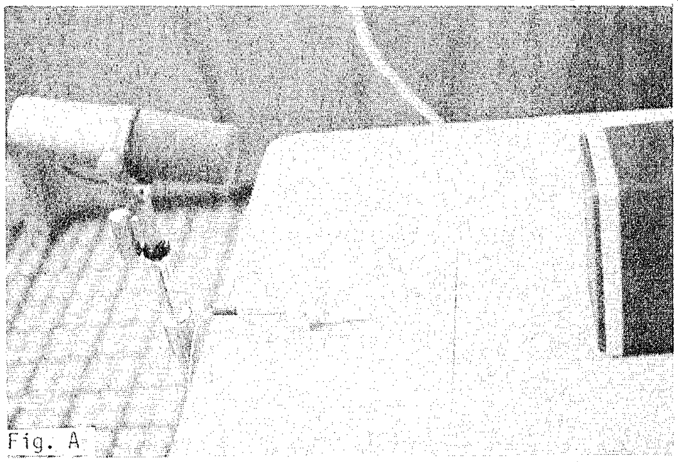


Fig. A

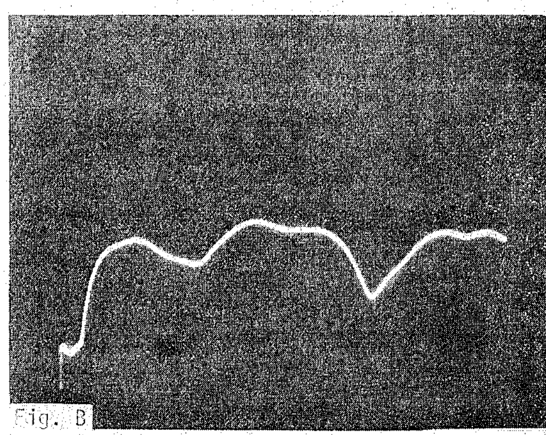


Fig. B

Figure A shows the SM-7 about 6" above a table top and Figure B shows the response from 0 to 10,000 Hz (linear frequency scale). Figure A also shows the PZM™ system on the table below the SM-7.

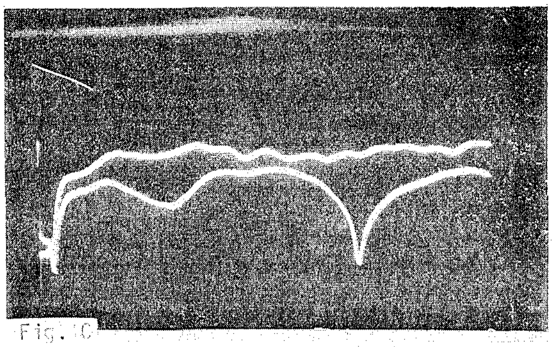


Fig. C

Figure C compares the response of the PZM system to the SM-7 under these circumstances.

If at this point you were to lower the SM-7 until it was against the surface of the table, it then smooths out in the 0-10,000 Hz region (the first dip moves out to around 20,000Hz)

What is startling is that if you now lift the test loudspeaker above the table and point it at the SM-7, the response anomolies again appear as deep notches *whereas the PZM system ignores them.*

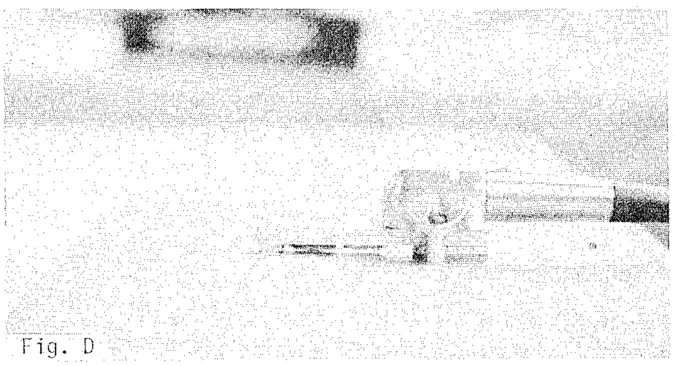
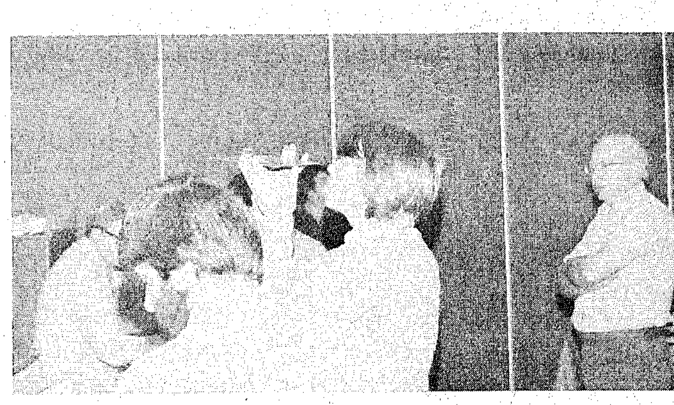


Fig. D



Figures D and E show various views of the PZM system.

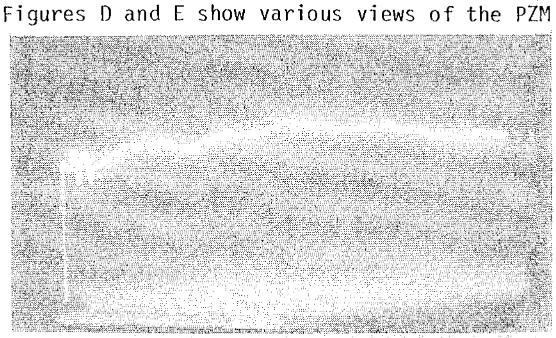


Fig. F

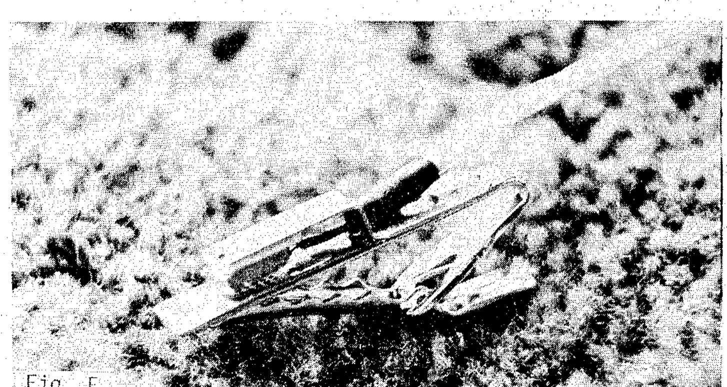


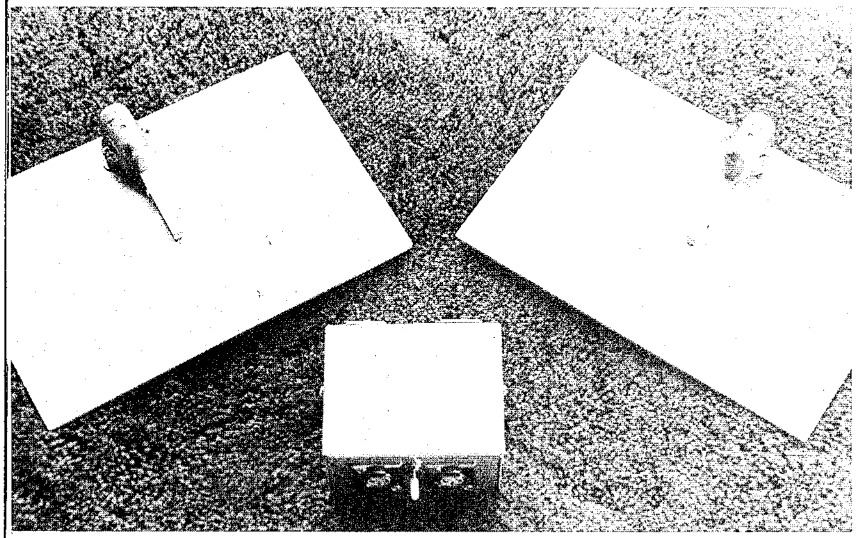
Fig. E

Figure F shows the PZM system's response from 0 to 20,000 Hz (our test loudspeaker's response drops off at 25,000Hz).

Tests made in the field indicate that conventional microphones will in the future be used only for "effects" and special "coloration" that performers devise and desire, and that all recording desiring accurate, uncolored, and calibrated response will use the PZM system.

That's about as revolutionary a change as we've encountered in our many years of audio.

PRESSURE ZONE MICROPHONE



MODELS AVAILABLE

"A" WITH 6X9 PLATE
"C" WITH 4X4 PLATE

REQUIRES POWER SUPPLY AND TRANSFORMER TO MATCH LOW IMPEDANCE INPUTS.

3 OPTIONS:
BT BATTERY-TRANSFORMER SUPPLY
PT PHANTOM PWR-TRANSFORMER
PXT PHANTOM PWR-TRANSFORMERLESS USES OP AMP TO MATCH IMPEDANCE

PRICE: To Syn-Aud-Con Graduates; $200.00 per pair with power supply. Calif: Plus 6% Sales Tax.

WHAT IT IS AND DOES

Utilizing a pressure response electret module mounted facing a plate so that a pressure zone develops, the PZM™ microphone provides exceptional pickup and fidelity with a balance between direct and reverberant sound. This new microphone offers unusual clarity and presence in stereo and multi-channel recording.

The PZM system has been used for piano, band section, solo and vocal pickup with excited acceptance by mixers and producers.

The PZM system can be used for conference table reinforcement, pulpits and many other applications in sound reinforcement.

Individual microphones also available.

WAHRENBROCK SOUND ASSOCIATES, 9609 CHEDDAR ST., DOWNEY, CA., 90242.

SEND ME ___ PZM MICROPHONE SYSTEMS.

MODEL _____ POWER SUPPLY _______.

NAME: ____________________

ADDRESS: ____________________

CITY: ____________________ STATE: ______

PHONE: AREA ______ NO. ______________ ZIP: _______

PRESSURE MEASUREMENTS IN PRIOR YEARS

After seeing our PZM^T.M.^ microphones in operation at the Los Angeles AES section meeting, Ludwig Sepmeyer, a well known acoustical consultant on the west coast, referred us to Beranek's 1949 edition of Acoustical Measurements wherein the work of A. London of the National Bureau of Standards wrote in 1941:

> *The microphones are placed as close to the panel as possible without touching.....The fluctuations in sound pressure measured over the surface of the panel are less than those measured in the body of the receiving room.*

The effect of increased SPL at the panel is noted as well.

Beranek's discussion of London's work is found on pages 874-876 of Acoustic Measurements.

We are indebted to Mr. Sepmeyer for his recollection of this data and his reminder that thorough researchers often have fundamental work overlooked by the casual observer.

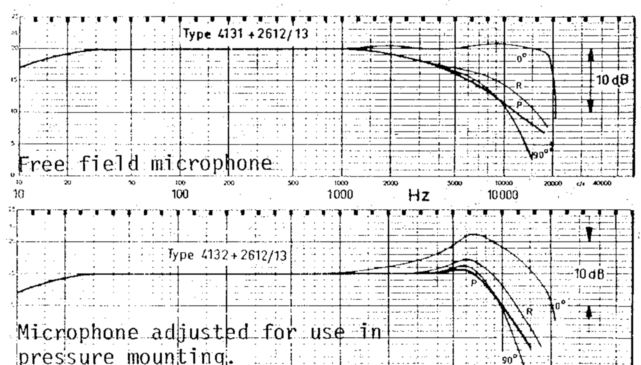
WHAT IS A "PRESSURE" MICROPHONE?

The question,"What is a 'pressure' microphone?" can have several answers depending upon the source you refer to. Beranek and Kinsler and Frey describe microphones whose output is proportional to sound pressure as *pressure* microphones. If the output is proportional to the pressure gradient, then they are called pressure gradient microphones (ribbon microphones, for example).

Bruel and Kjaer on the other hand refer to a microphone with a pressure response which is designed to be uniform (flat) as a *pressure microphone* and a microphone which is designed to be uniform (flat) for a 0° incidence free field response as a free field microphone.

The difference between these two types is that a Bruel and Kjaer free field microphone is damped in order to compensate for the pressure increase, which occurs at high frequencies, due to diffraction.

A Pressure Zone Microphone™, PZM, insures that no buildup of pressure can occur on axis but that only a typical cavity response (uniform in amplitude and phase over the entire diaphragm) can occur. Ed Long and Ron Wickersham were the first to point this out to us in our 1978 San Francisco class.

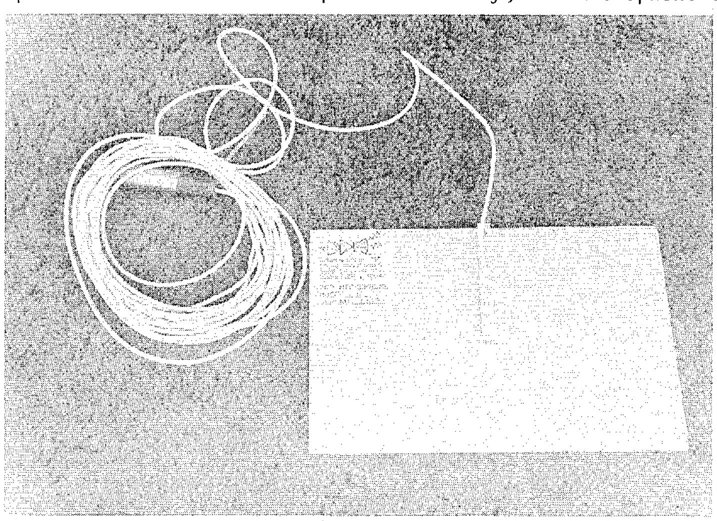


Examples shown here are ½" B&K microphone

Where normally the free field response would be the choice, because of the danger of error from on-axis signals into a pressure microphone in a random field, the PZM system insures a predictable acoustic field for which the microphone can be reliably calibrated without fear of undetected or unsuspected variations from strong reflections.

(Ed Long and Ron Wickersham have applied for a patent to cover their work with the pressure response pickup, PRP. Ken Wahrenbrock, who builds the PZM system shown below, pays a licensing fee to Long and Wickersham.)

A new form of pressure mounting is shown in Figure 2. This mounting technique employs a highly reflective metal plate as the bottom of a pressure cavity, the microphone diaphragm as the top, and the air itself as the sides.



The advantage of this arrangement is that there can be no response anomolies generated between the direct sound and a reflected sound *caused by the microphone's presence.*

The particular PZM system shown has exceptional transient response capabilities due to its extremely small size and wide range response.

In the arrangement illustrated it is obvious that no 0° incidence signal can arrive at the diaphragm. Thus this arrangement insures that the pressure response calibration of the microphone is never violated by the received sound field.

MEASURING ACOUSTIC ABSORPTION "IN SITU"

Using Time Delay Spectrometry (TDS) to measure the absorption coefficient of materials on, near, or of surfaces, is a simple variation of measuring the spectrum of a reflection from a hard surface.

It is convenient to work at distances greater than the lowest wavelength of interest (usually 4 to 10 feet) from the boundary surface used as a reflector - reference. A good stiff piece of 3/4" plywood serves as a portable reflector and for serious permanent facilities a dense concrete wall covered with a marble facing would make an excellent reference.

Microphone Requirements

At the distances normally used, the direct sound has passed the microphone, traveled to the boundary and been reflected back to the microphone. Since the tracking filter does not arrive "in tune" until the reflection arrives at the microphone this normally gives the reflection about 20 dB of S/N over the direct sound which was that far down on the filter skirts as the direct sound passed the microphone before being reflected.

When high absorption is present the reflection may now be down close to 20 dB, thus allowing the direct sound to interfere as both reflected and direct sound levels are near equality. This is, fortunately, easily handled by using a high quality cardioid microphone with at least 20 dB front-to-back discrimination. We use the Shure SM-7 because of its extremely smooth frontal response and the similarity between its front and back response.

Comb Filter Effects

Using TDS for these measurements has revealed that the surface of the absorbing material quite often reflects a signal *that is equal in level* to the signal sent back through the absorption material from the reference boundary. Being at substantially the same level, a "comb filter" effect results.

I believe that this comb filter effect has been overlooked to a large degree and could bear investigation especially as to the psychoacoustic importance to listeners in environments using absorption spaced out from boundaries in order to obtain enhanced low frequency absorption coefficients.

TDS, used at the time of material installation, is an unbelievably useful tool for geometric adjustment of reflectors and an instant confirmation of optimum absorption characteristics in situ. Especially interesting is the ability to watch the entire reflected spectrum as the boundary (or source) is rotated to allow study of the effect of angle of incidence on a sample.

It's time we moved a majority of measurements out of the laboratory and into the "real life" environment. The currently available analyzers are accurate, rugged, and easy to use.

ANECHOIC CHAMBERS VS TDS

To an engineer familiar with anechoic chambers, it is at first difficult to recognize that TDS is not only a way to obtain anechoic (without echo) measurements for less expense but that TDS is also a more accurate technique for these measurements.

The particular analyzer we are currently using for TDS measurements has I.F. filters with a "shape factor" of 10:1. This means that its filter "skirts" are 10 times wider at -60 dB than they are at -3 dB points. Since we typically use an I.F. filter bandwidth of 10 Hz, this means that at the -60 dB point the "skirt" width is 100 Hz. If we use a sweep rate, S.R. = 5000 Hz/sec, this then means that by placing our microphone

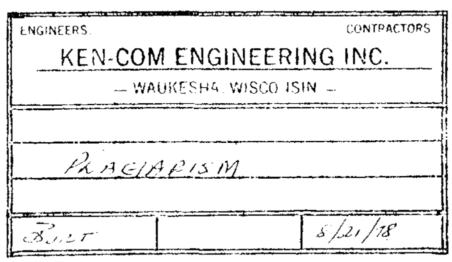
$$\text{Dist} = \frac{\text{F.O.}}{\text{S.R.}}(\text{vel of sound}) \qquad \text{Where F.O. = frequency offset of tracking filter and oscillator}$$

$$\frac{50\ \text{Hz}}{5000}(1130) = 11.3'$$

11.3 feet away from the nearest reflecting surface we have <u>60 dB of discrimination</u> against any reflection from that surface. At a S.R. = 10,000 Hz/sec (a realizable rate)

$$\frac{50}{10{,}000}(1130) = 5.65'$$

In an anechoic chamber which absorbs 99% of the energy arriving at its boundary, you are only down 20 dB. This means that TDS can and does literally provide 40 dB *more* discrimination than an anechoic chamber 100 times the TDS analyzer's cost.



NATIONAL BUREAU OF STANDARDS SOUND DIVISION DISBANDING?

Word has come to Syn-Aud-Con from several sources that the U.S. Department of Commerce is making quiet, secret plans to disband its Sound Laboratory staff and seriously curtail its research and standard setting activities in the areas of noise and hearing.

The reason offered for these plans is to "free up money for the pet projects of high ranking Commerce Department officials".

It is evident to Syn-Aud-Con that part of NBS' troubles stems from some very ill-advised steps into OSHA-type noise research with which a majority of Americans are fed up.

In a letter from a source we respect and feel is reliable, we are told "Persons in the Bureau (NBS) have been told that up to sixteen other activities may suffer similar demise in the near future, possibly including the division that works on basic electrical measurements. The potential disaster and chaos in all measurement standards is hard to comprehend. Our basic sound pressure measurements, microphone calibrations, etc., all tie back to NBS and their facilities. If the activities and training ground for acquiring the unique experience and attitudes necessary for maintaining these basic standards are done away with, the NBS traceability will soon become a farce."

Two points stand out in this potential mess:

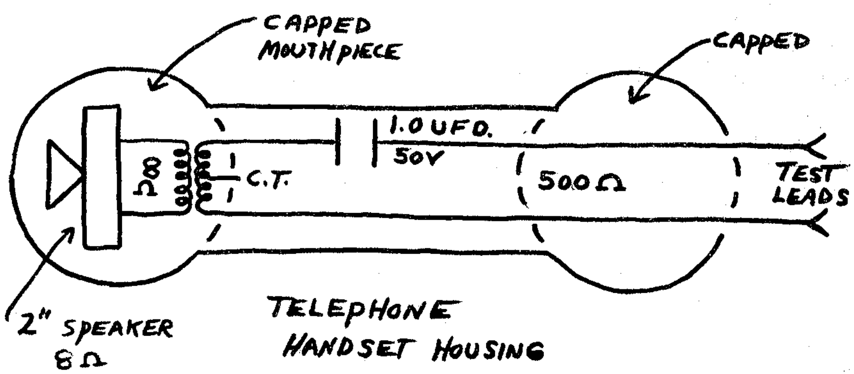
1. Some administrative officials allowed the prestige of NBS to be used to promulgate highly questionable research on EPA's (OSHA) behalf.
2. Instead of removing such appointed officials and seeing someone sensible is appointed in their place, the useful part of the activity is to be sacrificed to save the incompetents.

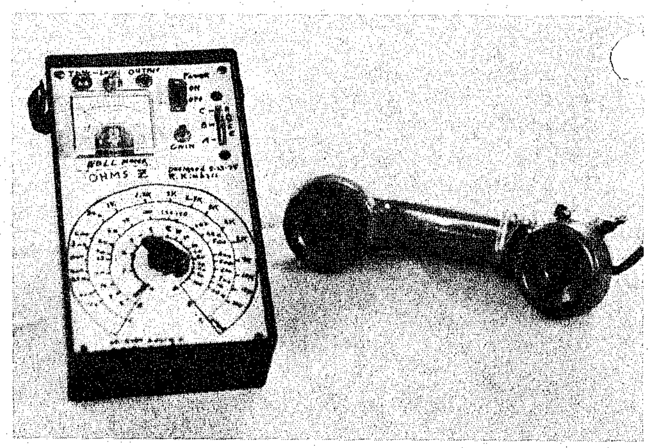
If you wish to register your protest, and you should, write or call Senator Adlai Stevenson (D-Illinois) Chairman of the Senate Subcommittee on Science, Technology and Space; and Juanita Kreps, Secretary of Commerce.

This ill-considered cut-back will have severe impact on the electronics and acoustics industry.

TELEPHONE HAND TESTSET

ROBERT KIMBALL in our St. Louis-Rauland class loves black boxes. He brought with him to class his home made, hand held test set, which led to a fruitful discussion of these versatile and helpful signal tracers. When GREG GARRY from Sioux Sound in Sioux Falls and CRAIG THOMPSON of Mueller Electronics in Peoria got home, they each sent us literature from Music Supply Company (PO Box 4080, Dallas) who make the Model TS-1 TesTSet commercially available for $49 - as well as a host of useful items for the sound contractor.





We took a picture of BOB KIMBALL's test set and he made a sketch and wrote up the following material:

Telephone handset used for test speaker. The telephone handset modified - see drawing - will

- Listen to a 70v speaker line
- Listen to a 25v speaker line
- Listen to a voice coil impedance circuit
- Listen to the input of a power amp
- Listen to the output of a tuner

The 600 ohm impedance with dc blocking in test lead makes this test speaker a handy test device around sound systems to check for signal or no signal conditions. (setting levels with this unit is not possible)

This device will monitor a telephone line without either party knowing of monitoring. Caution: Do NOT hold speaker up to ear when connecting leads. (Bob Kimball, City Electronics, 1206 E. 8th St., KC, Mo. 64106)

OPEN CIRCUIT POTENTIAL

According to the IEEE Standard Dictionary of Electrical and Electronics terms "open circuit potential" is defined: The measured potential of a cell from which *no current flows* in an external circuit.

When current is not flowing there *can be no power* since W = EI and all other convolutions such as $W = \frac{E^2}{R}$ presume current flow.

continued on the next page

Open Circuit Potential, continued

This is the reason for presuming a matched case when calculating available power (or potential power) for an open circuit voltage rating of microphones. The matched case would result in half the voltage appearing across the load resistor and half across the source resistance. This is the reason that 6 dB is deducted from the dBm calculation made by using the open circuit voltage in the equation for available power from the microphone.

$$\text{Microphone rating in dBm} = 10 \log \left(\frac{E_o^2}{.001Z}\right) - 6 \text{ dB}$$

where Z is the microphone's impedance

In *Sound System Engineering*, page 31, this is discussed and the point is made that microphones normally are used as voltage devices and not matched.

MEL SPRINKLE has written us recently pointing out that this distinction regarding open circuit voltages would be worthwhile emphasizing.

IMPORTANT CORRECTIONS TO EARLIER NEWSLETTERS AND TECH TOPICS

On Page 28 of the January 1978, Volume 5, No. 2 Newsletter, a very useful and important equation was miswritten:

Incorrect $$D_x = \frac{D_{2ss}}{10^{\left(\frac{Me}{20}\right)}}$$

Should have been written $$D_x = \frac{D_{2ss}}{\sqrt{Me}}$$

On Page 23 of the July 1978 Newsletter, Volume 5, Number 4

Now reads: $\sqrt{D}$ is the energy density in watt seconds per cubic meter

Should read: D is the energy density in watt seconds per cubic meter

Additions to this page include:

$\sqrt{D}$ = particle velocity in M/sec

$\sqrt{D} = \sqrt{1a/Ra}$

$D = \frac{P^2}{Ra^2}$ Ra = pc

$Z = \frac{P}{S\sqrt{D}}$

On Page 5 of Tech Topic Vol 5, Number 13, "Speech Reception and Information", the instructions for entering data and its subsequent manipulation now reads:

Enter both sides of card

Enter L_D	Press "A"	
Enter L_R	Press R/S	
Enter L_N	Press R/S	Incorrect
Enter T	Press R/S	

and

A	B	C	D	E
	L_D	L_R	L_N	T

Should read:

Enter both sides of card

Enter dB_D	Press "A"	
Enter dB_R	Press R/S	Correct
Enter dB_N	Press R/S	
Enter T	Press R/S	

and

A	B	C	D	E
	dB_D	dB_R	dB_N	T

On Page 3, Example # 3, of Tech Topic Volume 5, Number 13, "The dB and dBm Revisited," contained a miswritten question to a correct answer.

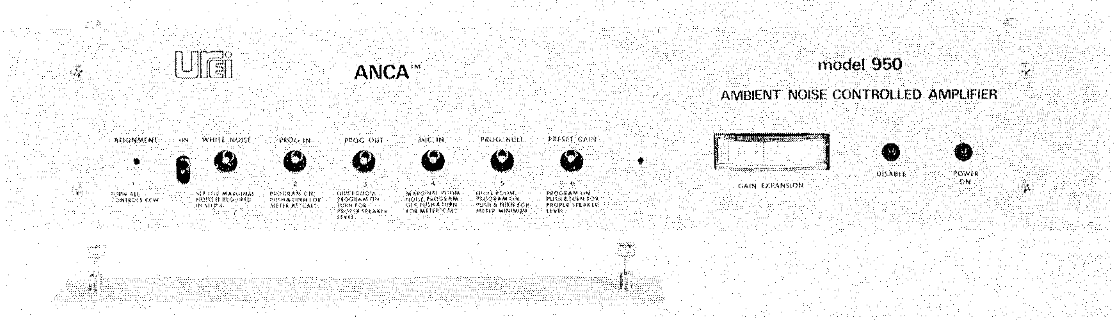
Now reads: "Suppose that across 600Ω....."

It should have read:

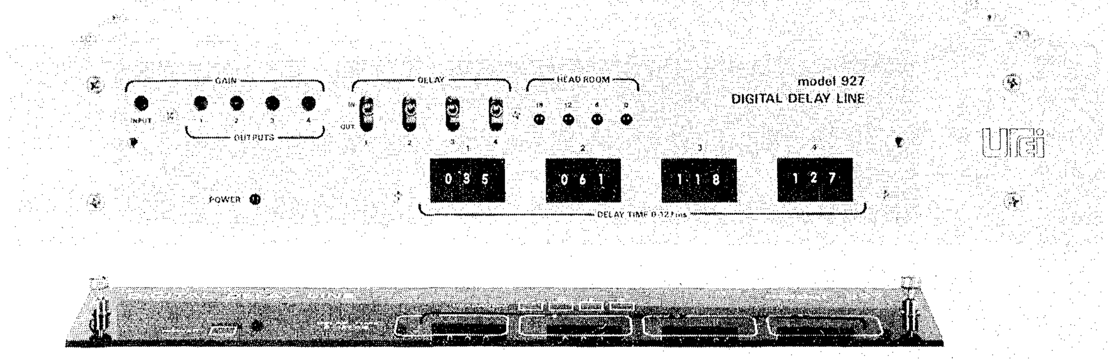
"Suppose that you read an apparent +18 dBm (6.16V) and that you then measure an actual +12 dBm across the unknown load. What impedance is the unknown load?"

NEW PRODUCTS FROM U.R.E.I.

This issue of the Newsletter truly illustrates the joy of being in the audio business today. The revolution we have been telling you about during the past six years is now in full swing. Just glance through this issue and then take note that we only had room to mention about 1/10th of the new things going on in the marketplace. Two very useful new items are from U.R.E.I.

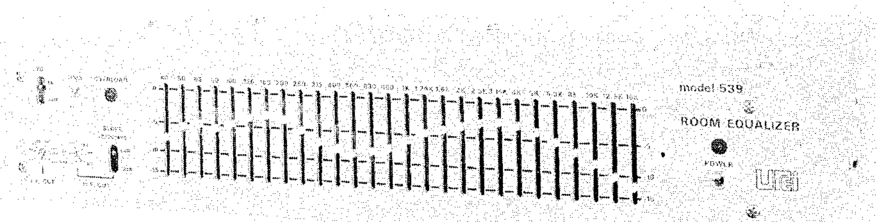


The ANCA - Ambient Noise Controlled Amplifier - automatically controls the level of the sound system as the ambient noise level changes, *even during announcements*. At long last a really practical form of this type of dynamic volume control amplifier.



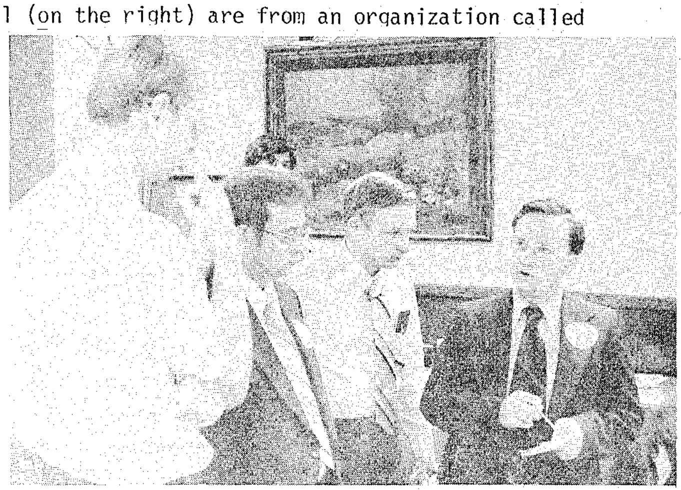
U.R.E.I. has also brought out a digital delay line with four delays adjustable in one millisecond steps out to 127 msec. Add to these specs, 92 dB S/N and 12KHz full power bandwidth and the picture begins to come into focus.

We have used the new U.R.E.I. model 539 1/3-octave equalizer in some five classes and we quite willingly rate it as first choice in active equalizers for the commercial sound market.



ENERGY IN THE TERM SYNERGY

The four audio engineers pictured here, headed by John Goodell (on the right) are from an organization called The Institute in Basic Youth Conflicts. They are specialists in setting up entire arenas for large audiences for the seminar leader, Bill Gotherd, who is generally presented by video projection and reproduced sound. (We plan to attend a seminar in San Diego in February.)



John brought his group into the Chicago class for training and interchange of information.

John is an excellent engineer and really has his basics in hand. It's men like John that put energy into the term synergy.

CASSETTES OF SPECIAL GRADUATE MEETING

Having now listened repeatedly to the tape cassettes of the special graduate meeting last April, I can only say that those of you without a copy are missing an aural view of where our industry not only is, but where it's headed. Futurology is not practiced that frequently in audio and it is almost intensely present in these tapes.

We do not plan to re-duplicate these priceless tapes once our present supply is exhausted and they should therefore have increasing value as the years pass. Think back and realize what tapes of Wente and Thuras; Steinberg and Snow; Sivian, Dunn and White; Fletcher and others would be worth today had they been recorded in the 1930s.

What would you pay if you could have had a tape of Einstein giving one of his early papers and translate that to having a tape of Heyser forty years from now. That's Syn-Aud-Con's view of these tapes. The men on these tapes are not in their formal paper-giving mood but are freewheeling it with an attentive, respectful audience. Heyser with his hair down; Peutz in his most generous, sharing mood; Moir playing the devil's advocate with devotion, skill and wit; and John Hilliard giving peeks under the curtain at our industry's beginnings -- as we have said, unique material to say the least.

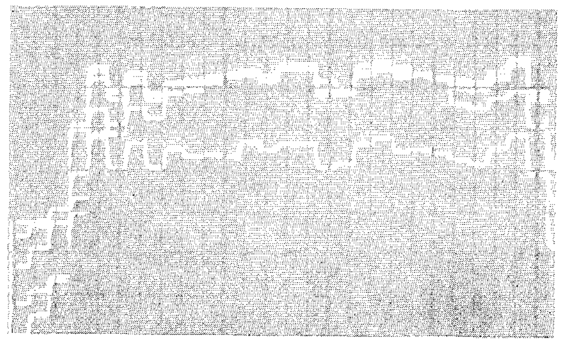
ANSWER TO dB QUESTION

This can not be determined without knowing the input and output impedances. For example: If the input Z = 100Ω and the load Z = 10Ω, then the device has 4 dB of gain. If the input Z = load Z, then the device has a 6 dB loss.

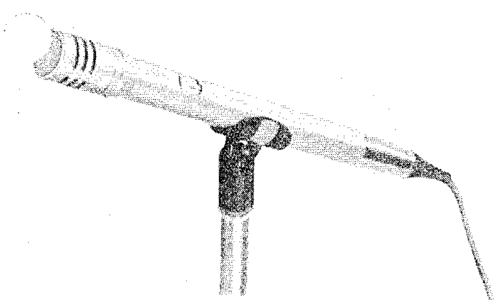
THE SHURE SM81 MICROPHONE

The Shure SM 81 unidirectional condenser microphone is designed for professional applications in recording studios. To our knowledge this is the first condenser microphone that Shure has ever offered.

Our tests of this unit during the San Francisco class shows an extremely smooth frequency response both front and back and very clean sound.



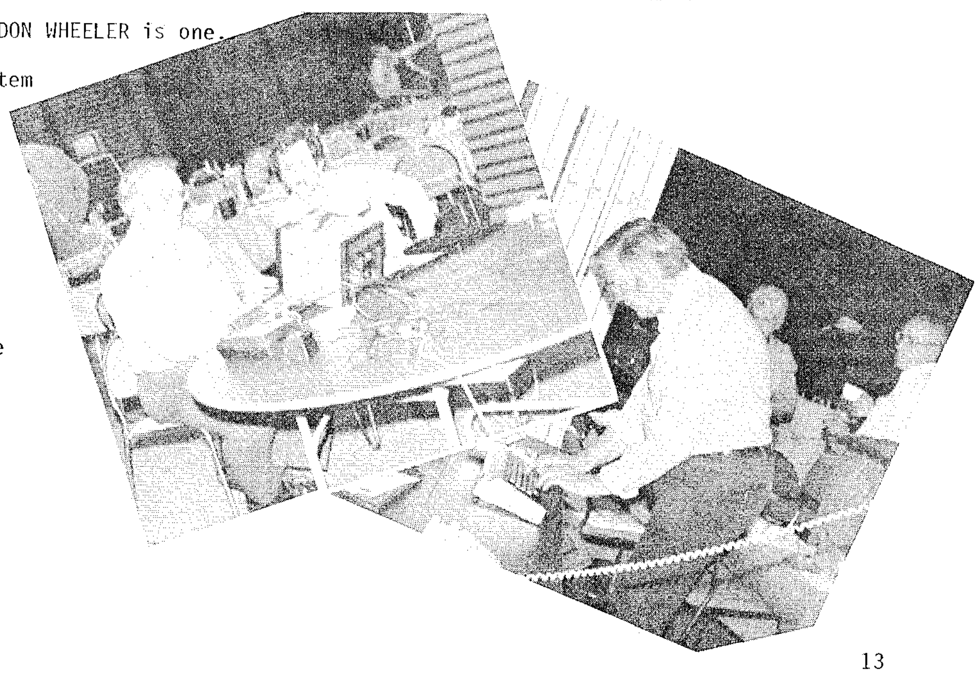
We have not yet had a chance to try a pair of these units on a musical group but suspect that this cartridge contains the seed for an exceptional PZM capsule somewhere in the future (in an omni-directional pressure calibrated format, of course). Shure has designed two power supplies for use with the SM 81.



Shure specification sheets are a particular pleasure to read because they are complete, accurate, use the dB correctly (dBV, for example, is referenced to 1 volt and is not presented as anything other than open circuit voltage level and are verifiable in field tests. They give both rated 150Ω and actual 85Ω impedances and their response curves are believable.

SOUND SYSTEM DESIGN PROGRAM IN BASIC

There is a new breed of Computerholics. DON WHEELER is one.



If you would like to obtain the sound system design program in basic, write:

Don Wheeler
Alpha Electronics
623 Sligo Ave
Silver Spring
MD 20910

Don also has many more programs available on both audio and financial. He would like to recover some of his costs so write for a listing of programs available and their cost. And if you have programs, let him know. It may be you can swap programs.

SPECIAL CLASS FOR RAULAND DISTRIBUTORS
ST. LOUIS, SEPTEMBER, 1978

WASHINGTON, D.C. CLASS, 1978

The old (an 1886 graphaphone by Columbia) Edison type cylinder phonograph being measured by the new (Ivie 1/3-octave real time analyzer) were both given Judy's approval. Judy's ears indicate a slight reservation about the price Carolyn paid at a sidewalk antique sale in Gettysburg to make this kind of noise. "That's a lot of Kal Kan" I think Judy said.

REPORT ON AUTOMATIC MIXERS

The Dugan mixer design bought by Altec was the first commercially marketed automatic mixer to incorporate the Number of Open Microphones Attenuator (NOMA) idea first suggested to me in 1968 by KEN PATTERSON of Kansas City. Ken had built a manual version in a switch that attenuated the output of a mixer according to my NOM equation as the number of inputs to it were increased.

Since, to my knowledge, I was the first to write down the NOM equation, though others had observed and commented on the effect without expressing it mathematically, I have been an interested observer of each stage of development that automatic mixers have undergone.

The Dugan mixer incorporated a number of innovations beyond merely controlling the mixer's output level, and as hand built by Dan Dugan, worked very well in council chambers, churches, and board rooms. The Altec version ran into initial serious quality control problems as well as their historic tendency to have difficulties with an "A" model.

The Rauland Mixer

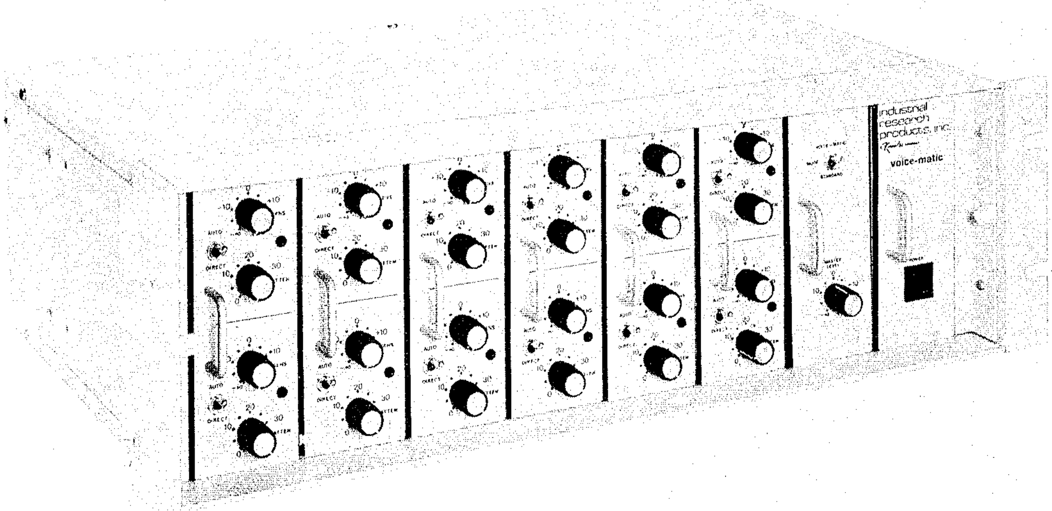
The Rauland Corporation has offered its distribution an automatic mixer that, after initial trial productions, is now in its field tests. Our first encounters with the 3535 Log 8 promises a happy future for it. (See next page for more detail)

The Industrial Research Products Mixer

Industrial Research Model DE-4013 Microphone Mixer is the latest entry in the improvement of the idea. The DE-4013 incorporates a number of worthwhile innovations.

Dynamic Threshold Sensing (DTS) is used to differentiate between active and inactive microphones. Every ten milliseconds or less a threshold scan starts at high levels and moves to low levels. All microphone inputs are scanned simultaneously. When the threshold level decreases to the channel having highest level, that channel is given temporary "channel on" status and simultaneously the threshold is set high and another scan initiated. On a repeated scan the same or a different channel may have a high level and receive temporary "channel on" status. In this way, the active microphone channels are detected and updated to maintain "channel on" status.

Several microphones can have simultaneous "channel on" status and be effectively updated without dropouts. The master amplifier gain is decreased by 3 dB for each doubling of the number of simultaneously active microphones. In systems with very large numbers of microphones,a large master attenuation range is not needed to protect against simultaneous activation of all microphones. The maximum number of microphones which can be effectively updated is limited as the mixer gradually shifts to a time share mode to prevent simultaneous activation of all inputs. This assures absolute stability for an unlimited number of microphones in the system.



Additional capabilities include disabling the automatic feature on any input not having an open microphone such as a tape recorder, phono, or tuner input; priority assignment to any input along with channel status indication; and up to 12 plug in inputs per chassis with indefinite linkage of chassis to each other.

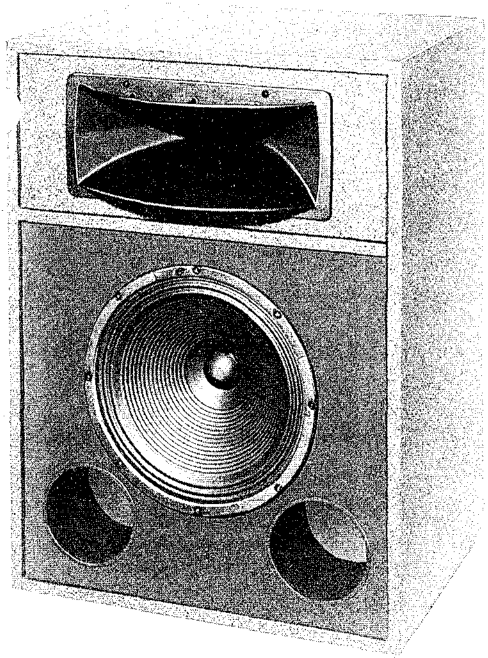
Conclusion

Automatic microphone mixing is now available to any audio professional ready to utilize this most useful tool. In our opinion, failing to use one of these units where multi-microphone use is anticipated and an operator is not present (which describes most church systems) is to make your installation obsolete before it is delivered to the customer.

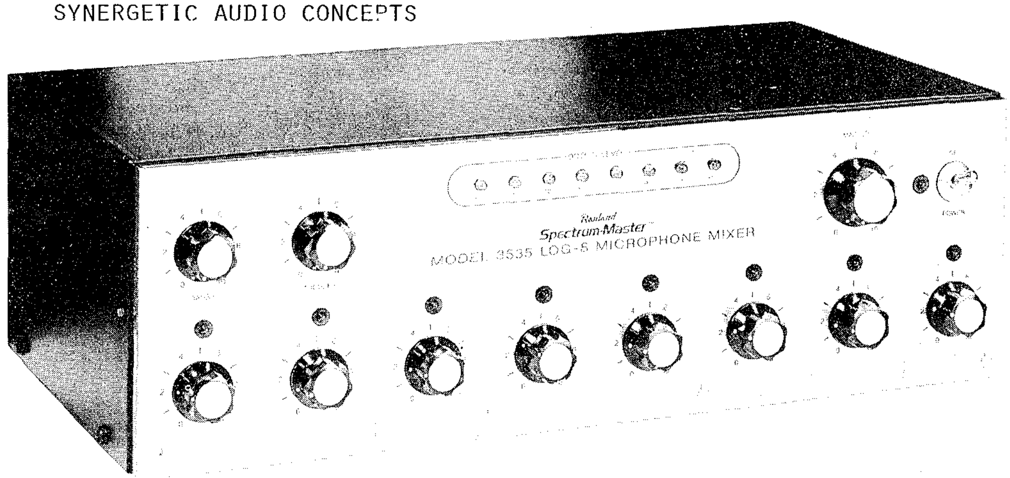
NEW PRODUCTS FROM RAULAND-BORG CORPORATION

Rauland-Borg Corporation's extensive product line has at least two new products of fundamental interest to Syn-Aud-Con graduates. The first is a very rugged, wide range, high performance loudspeaker for commercial sound use. Using a 500 Hz crossover from the 15" woofer to the Emilar horn and driver, it is not hard to accept its 100 watt power rating (average power--calculated from RMS voltage and the impedance). The model # is MLS-5.

As you have all witnessed in class, the Rauland Engineering personnel know how to design loudspeaker systems and our measurements of this system have made it Syn-Aud-Con's first choice for the category it serves. We consider it ideal for small theaters, auditoriums, discos, churches, etc. (continued next page)



MLS-5 Professional 2-way Speaker System



3535 Log-8 Microphone Mixer

A second Rauland item that will compete with the Altec and Industrial Research Product automatic mixers is the Rauland Model 3535 Log-8 microphone mixer.

As has been the case with equalization, Rauland engineers have made sure that their distributors can compete effectively in any market place.

Syn-Aud-Con has been conducting special classes for Rauland contractors and they confirm our opinion that Rauland is today the leading line available to a contractor desiring to handle a full line from one manufacturer. While the older leaders stumble about in the market place, Rauland has set a course that has sailed them past and into the lead. Ten years from now many of you will wish you had read this observation more carefully.

1/3 SCALE MODELS OF LOUDSPEAKERS AVAILABLE

Barry McKinnon builds 1/3 scale models of a majority of well known commercially used speaker enclosures, including Altec, JBL and Gauss.

They are of an optimum size to demonstrate comparative enclosure bulk, the concepts of cluster design, cluster density, positioning, etc.

You can write Barry McKinnon at 2828 - 11th Ave SE, Calgary, Alta, T2A 0E7 (Phone 403-273-7895) if you are interested.

★ ★ ★ ★ ★ ★

Fans come first when Kiss plays

By William J. Cahill

NEW YORK (UPI) — Maybe the glitter-rock band Kiss could change its name to "Smack."

Kiss, they of the painted faces and outrageous costumes, smacked a full house at Madison Square Garden recently with something close to 180 decibels and outrageous visual effects which were eaten up by the teen-age "Kiss Army" fans.

Those over the age of 21 seemed to have a "what-am-I-doing-here" look on their faces.

Kiss opened with "I Stole Your Love," a driving foot-stomping number that may have sounded better than it was because the band kept the opening night audience waiting more than an hour past the billed starting time.

The news article sent in by MARK ROGERS of InterMountain Sound in Boise, Idaho reveals again that only mentalities that have already been subjected to 180 dB levels would believe what they read in a news source.

180 dB-SPL at the ear is an intensity of

$$10^{\left(\frac{180}{10}\right)} \times 10^{-16} \text{ watts/cm}^2 = 100 \text{ acoustic watts/cm}^2$$

That's not only beyond the threshold of pain, but of mortality. There is a difference between exciting fans and incinerating them.

DOPPLER RADAR "MINI-COURSE"

JOHN PHELAN, Manager of Professional Sound Products at Shure Brothers Inc., submitted a carefully detailed analysis of the current ECM (electronic counter measures) being used against police radar.

If you, at first blush, feel that "If I don't speed, why should I worry about radar?" read *Helping the Speeders Beat the Radar Rap* in the IEEE Spectrum, August 1978.

In the IEEE article the following carefully researched set of comments is made:

> *Law abiding motorists typically reason they never speed, so they have nothing to fear from radar - an attitude that assumes a high level of infallibility on the part of enforcement agencies and their equipment. However, because many policemen do not understand how radar works, there is ample opportunity for innocent motorists to be ticketed . Much of this problem revolves around target identification. ...The officer must carefully and constantly observe all traffic while monitoring the radar unit since... the strongest reflected signal received at any given moment and the speed reported is not necessarily that of the nearest vehicle.*
>
> *Radar readings can be affected by ignition noise, CB transmissions, and other nearby RF energy fields. Also some radars have calibrate, reset, and other control buttons that can be manipulated to provide any desired readout, regardless of actual vehicle speeds being observed - which might be used by some municipalities to generate easy revenues from motorists passing through and are not in much of a position to complain.*

The International Association of Chiefs of Police has recognized the problem and has asked the National Bureau of Standards to help establish minimum standards for police radar. It has been our observation that when a law is passed that the majority really is against, they won't obey it. The current 55 mph speed limit is clearly such a law.

JOHN PHELAN's article can give you some ammunition if your community is using radar for revenue rather than reduction of speed.

It's good to see that someone else is interested in the various and sundry ways in which the local arm of the law keeps tab on citizens' speeds (Radar Traps and the Doppler Effect, Syn-Aud-Con Newsletter, Vol. 5, Issue 3, April, 1978). However, Mr. Symmes failed to carry his calculations to the correct conclusion.

The formula he used, $F_1 = F_0\left(\frac{C \pm V}{C}\right)$ is correct for doppler shift in the case of the train whistle--that is, where the transmitter is moving (train whistle) and the receiver is stationary (your car). Police radar takes a slightly different path. Since the transmitter and receiver are both in the same place, whether they are moving or not is of little consequence. What is important is that the signal must go out from and return to the same place; i.e., the radar unit. Therefore, it has a double path to follow--out from the radar, reflect off the target, and back to the radar. This calls for a new formula to take into account the double wave path. The formula is

$$F = 2\left(\frac{VF_c}{C}\right)$$

Where F is the doppler shift in hertz
V is the vehicle speed of target
F_c is the frequency of radar carrier in hertz
C is the speed of light (186,273 miles per sec)

Note: V and C must be in the same unit of measure--mph, ft/sec, etc.

The formula holds true whether you are coming at or going away from the radar. As can be seen from the formula, the amount of doppler shift for a given speed will change depending on the carrier frequency being transmitted. For X band radar (10.525 gigahertz), the shift is 31.4 Hz/mile/hour. For K band (24.150 gigahertz), the shift is 72.13 Hz/mile/hour. As an interesting side note, a moving radar cannot clock a vehicle when the police car speed is greater than the vehicle speed. In other words, the law cannot zip up behind you and read your speed without first bringing his patrol car to a speed lower than your vehicle, plenty of time to slow down.

But what does all this mean? To make it easy to understand, let's take Bill's case of 60 mph. You would like the police radar to read 55 mph. First we need to know what the radar would normally see at 55mph (with X band radar).

So: $F = 2\left(\frac{VF_c}{C}\right)$

F = 1729.02 Hz shift for 55 mph
Total = 10.525 gHz + 1729.02 Hz = 10,525,001,729.02 Hz

Now, let's see what we have to send to make the radar "see" 55 mph. Back to Bill's formula, since there is only one wave path, from the car to the radar.

$$F_1 = F_0\left(\frac{C \pm V}{C}\right)$$

If we sent a signal of 10,525,001,729 out from our car, at the 60 mph speed, the doppler shift of our 60 mph speed would have to be added to that signal to arrive at what the police radar would actually "see". Thus the police radar in this case would "see" 10,525,002,676 Hz or 85.2 mph (31.4 Hz/mile/hour). Not too cool. What has to be done is adjust the transmitted "55" for the actual speed of 60 mph. Since the 60 mph speed adds a doppler shift of 947 Hz (single wave path), we must subtract that from the carrier frequency. So a transmitted carrier of 10,525,000,782 (10,525,001,729 Hz - 947 Hz) would have to be sent in order to fool the police radar into thinking you are going 55 when indeed you are going 60.

One more point. Don't forget that it only holds true for an on-axis radar shot. For every degree the radar

continued next page

Radar Traps, continued

is off-axis of your car, there is cosine error to contend with. The formula now becomes

$$F = 2\left(\frac{VF_c}{C}\right) \cos \phi \qquad \text{Where } \cos \phi \text{ is the cosine of the angle of the radar beam}$$

Suffice it to say that any cosine error reads in your favor up to 15°. After that, any respectable radar manufacturer will admit that the radar speed reading is practically useless as evidence.

If all this sounds rather mysterious and foreboding, I'm sure you will realize why most people do not fight radar tickets; however, a little knowledge can go a long way in the courtroom. As for Bill Symmes' transponder, a much simpler method would be broadband noise (pink?) covering all possible doppler shifts due to both actual speed and cosine error. Why fool the radar when you can render it temporarily useless?

If anyone would care to discuss this further, I can be reached at (312) 866-2522 or by mail c/o Shure Brothers Inc, 222 Hartrey Ave, Evanston, IL 60204.

THE IVIE REAL TIME ANALYZER

We have now had the opportunity to work with the new Ivie IE-30A 1/3-octave RTA in three classes.

We have found some unexpected, to us, virtues in the IE-30A as well as a caution.

Virtues First

During rapid sine wave sweeps of the sound system near regeneration, the IE-30A proved to be the easiest analyzer to spot "ringing" frequencies, even beating our over 4 times more expensive GR unit.

And for viewing at the rear of a classroom, the IE-30A remains visible at surprisingly long distances.

Caution

The LED matrix seems to be susceptible to including at least one burnt out LED after short useage. In fairness, it must be said that when this occurs graduates tell us Ivie breaks all records for servicing the unit and instantly returning it.

Ivie has come a long way toward achieving the complete portable test lab and we can only pray that they build a TDS analyzer this same size ready to plug into the NLS battery operated oscilloscope.

We'll be reporting further on this unit as we become more familiar with its various operating features. To date, our impression is that it is all it is claimed to be by Ivie. The IE-17A shown attached to the IE-30A in the photograph has yet to be actually produced for sale. When it is, we'll be using it and reporting on it.

SWINTEK WIRELESS MICROPHONES

During the May 1977 AES Convention in Los Angeles, Syn-Aud-Con was approached by Bill Swintek and asked if we would care to test a new wireless microphone system they had developed. After discussing with him our negative experiences with wireless units, we were surprised to find that he felt he could overcome our major objections.

Common problems with wireless microphones are:

1. Unwanted outside interference (C.B., pocket paging, or just plain intermittent strange noises)
2. Erratic and unstable behavior. A trained RF man can keep the unit tuned but not the ordinary user.
3. Poorly constructed
4. Major deterioration of the sound quality normally available from the microphone when used separately from the wireless system.

We have used the Swintek Mark IV-50 VHF unit in four Syn-Aud-Con classes - two of them at Airports (St. Louis Lambert Field and NYC La Guardia) where we were able to ascertain tremendous RF fields were present but unable to interfere with the Swintek system.

We did encounter some noise in the first two classes and early in the third class but it was identified as a bad connection in the microphone cord and not a system problem. We are very pleased to have found this unit thoroughly suitable for our Syn-Aud-Con use and that we are greatly enjoying the freedom it provides.

The microphone has the same response as a wireless as it has when used alone. The unit hasn't been tuned or adjusted in any way since we started using it; no interference has occurred, not even in one of the world's worst RFI areas, and other than a faulty microphone cable which caused static-like noises when accidentally bent the wrong way, the unit has exhibited exceptional ruggedness.

The Swintek literature seems to tell the truth and has proven useful in providing an understanding of their approach to the wireless problem.

If you want to be in touch with Swintek, call Bill Swintek in Sunnyvale, CA (408)245-8720

ARTICULATION LOSS CHARTS

ED LETHERT of Minneapolis has literally overwhelmed us with input from his output. We are in the process of preparing for future Newsletters computer programs that Ed has prepared on:

1. Ted Uzzle's loudspeaker coverage Tech Topic
2. A remarkable new look at improving the simple form of the acoustic gain formulas
3. A detailed breakdown of the individual parameters in Peutz's newest $\%Al_{cons}$ equations
4. A new computer program for utilizing Curt Enerson's distributed loudspeaker Tech Topic
5. A reactance/resonance program
6. A line loss program for solid, solid tinned, stranded, and stranded tinned copper wire
7. A computer program for solving N from Ed's own Tech Topic on the subject.

As if this wasn't enough to review and study (this Newsletter is put-together during our week break between the New York and Atlanta class -- by "put together" I mean, written, typed and paged), Ed has his larger computer printing out small books containing the $\%Al_{cons}$ for RT_{60} from 1.5 secs to 10 secs. Sample print outs from two pages of this book are shown here. (The sheets in the booklet will not be this small, but at least 1/4 larger print.)

RT60 = 1.5 SECONDS

SPLN:	-25	-20	-15	-10	-5	0
SPLR:						
-10	4.4	4.87	5.66	7.39	11.64	21.38
-8	4.79	5.35	6.2	7.86	11.77	20.94
-6	5.17	5.86	6.81	8.46	12.06	20.55
-4	5.52	6.35	7.45	9.16	12.54	20.29
-2	6.14	6.81	8.09	9.93	13.18	20.22
0	6.89	7.21	8.68	10.71	13.95	20.37
2	7.72	7.73	9.17	11.41	14.75	20.71
4	8.61	8.62	9.51	11.97	15.45	21.11
6	9.54	9.54	9.66	12.3	15.93	21.43
8	10.45	10.45	10.46	12.34	16.08	21.48
10	11.3	11.3	11.3	12.08	15.84	21.13

SPLR AND SPLN ARE RELATIVE TO SPLD

RT60 = 1.6 SECONDS

SPLN:	-25	-20	-15	-10	-5	0
SPLR:						
-10	4.46	4.92	5.71	7.45	11.71	21.46
-8	4.87	5.43	6.28	7.94	11.86	21.05
-6	5.28	5.96	6.92	8.56	12.19	20.7
-4	5.66	6.5	7.6	9.31	12.7	20.48
-2	6.33	7	8.29	10.13	13.4	20.47
0	7.12	7.45	8.94	10.97	14.23	20.68
2	8.01	8.02	9.49	11.75	15.1	21.09
4	8.98	8.98	9.9	12.38	15.89	21.59
6	9.98	9.98	10.11	12.79	16.47	22
8	10.97	10.97	10.98	12.9	16.7	22.15
10	11.88	11.89	11.89	12.69	16.53	21.89

SPLR AND SPLN ARE RELATIVE TO SPLD

RT60 = 1.7 SECONDS

SPLN:	-25	-20	-15	-10	-5	0
SPLR:						
-10	4.51	4.97	5.76	7.5	11.78	21.54
-8	4.94	5.5	6.35	8.01	11.95	21.16
-6	5.38	6.07	7.02	8.67	12.3	20.84
-4	5.8	6.64	7.74	9.45	12.85	20.66
-2	6.51	7.19	8.48	10.32	13.6	20.7
0	7.36	7.68	9.18	11.22	14.49	20.97
2	8.31	8.32	9.8	12.08	15.44	21.46
4	9.34	9.35	10.27	12.79	16.32	22.04
6	10.42	10.42	10.55	13.27	16.98	22.55
8	11.48	11.48	11.49	13.45	17.31	22.8
10	12.47	12.47	12.48	13.29	17.21	22.62

SPLR AND SPLN ARE RELATIVE TO SPLD

RT60 = 1.8 SECONDS

SPLN:	-25	-20	-15	-10	-5	0
SPLR:						
-10	4.56	5.02	5.81	7.56	11.84	21.61
-8	5.02	5.57	6.42	8.09	12.04	21.26
-6	5.48	6.17	7.11	8.77	12.42	20.98
-4	5.94	6.78	7.87	9.59	13	20.84
-2	6.69	7.37	8.66	10.51	13.79	20.92
0	7.59	7.92	9.42	11.47	14.75	21.26
2	8.59	8.6	10.1	12.39	15.77	21.81
4	9.7	9.7	10.64	13.18	16.73	22.48
6	10.85	10.85	10.98	13.74	17.49	23.08
8	11.99	11.99	12	13.99	17.9	23.43
10	13.05	13.05	13.06	13.89	17.87	23.34

SPLR AND SPLN ARE RELATIVE TO SPLD

Syn-Aud-Con would like to know how many of you would be interested in buying such books for distribution to architectural engineers, consultants, etc.? If sufficient numbers of you are interested we are sure the cost can be held to $5.00 per copy. Let us hear from you.

With this latest effort by Ed Lethert the pieces necessary to the development of a "Westinghouse Lighting" type handbook on sound systems for architects and engineers are falling into place. All the tools necessary to the construction of such a manual are individually at hand and need only to be brought together in a coordinated, coherent fashion as tables and charts rather than as equations.

MORE ON NEW PEUTZ EQUATION

From ED LETHERT, Minneapolis:

The new Peutz equation is included here in a form which shows certain features which must be known in order to program other than an HP calculator

$$A = -0.32 \log \left(\frac{L_R + L_N}{(10\ L_D) + L_R + L_N} \right)$$

$$A > 1 = 1$$

continued on the next page

Ed Lethert - Peutz programming, cont.

$$D = -0.32 \log \left[\frac{L_N}{(10\ L_R) + L_N}\right]$$

$$D > 1 = 1$$

$$C = -\left[\frac{\log\ (T/12)}{2}\right]$$

$$B = C \times D$$

$$Al_{cons} = \left[10^{(-2((A+B)\ -(AB)))}\right] + 0.015$$

People programming the TI 50 sometimes miss the comparative tests or the change of sign on the "C" part of the equation.

Here are the three equations that describe the chart relating Al_{cons} to signal-to-noise ratio. (Chart on Page 71, Fig. 4-22 of SSE)

$$Al_{cons} = \log 10^{\left[\frac{(\text{Log } 0.09T) \times (S/N + 10)}{35}\right]}$$

$$\%Al_{cons} = 100 \times Al_{cons}$$

$$S/N = \left[\frac{(\text{Log } Al) \times 35}{\text{Log } .09T}\right] - 10$$

$$RT_{60} = \frac{\left[\text{Log } 10^{\left[\frac{(\log Al) \times 35}{S/N + 10}\right]}\right]}{0.09}$$

I've been giving some thought to the equation that allows one to apply a S/N correction to Al_{cons} as calculated during design. The correction factor equation is:

$$C.F. = \left[\frac{1}{.09T}\right]^{\left[\frac{25-S/N}{35}\right]}$$

$$\%Al_{cons}\ (S/N < 25\ dB) = \%Al_{cons}\ (\text{calculated}) \times C.F.$$

INTERACTION BETWEEN SYSTEM PARAMETERS

ANDY SOBIERALSKI of Rocel in Pittsburg sent in the following print out from an HP 9821A computer.

This plot reveals several interactions between system parameters.

Wouldn't it be interesting to see iso-articulations plotted for varying positions in a room, for example, or desired Q by seat, thus giving us the shape of a perfect polar pattern for that room.

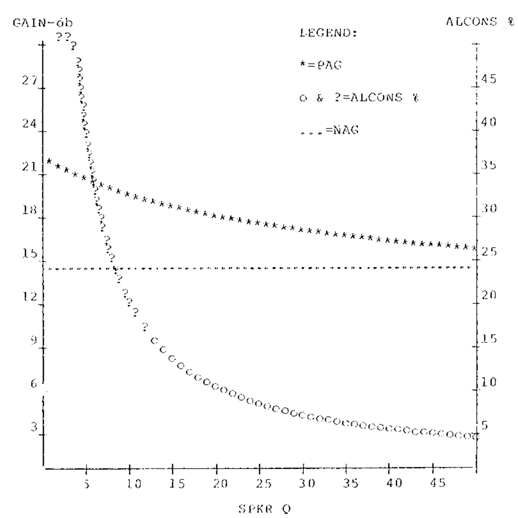
All attempts at plotting are beneficial and we look forward to seeing others. In the meantime, our thanks to Andy for having led the way.

Q vs PAG AND ALCONS %

PROJECT.....................

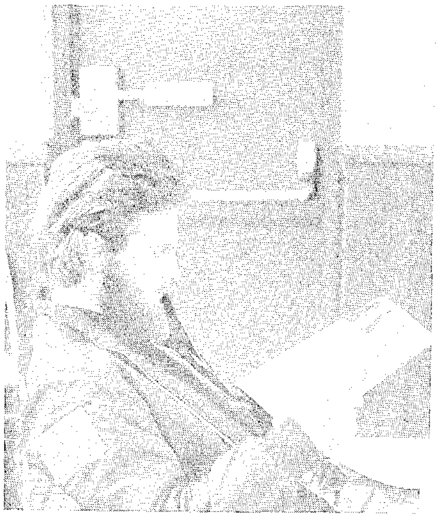
DATE............ REMARKS.....................

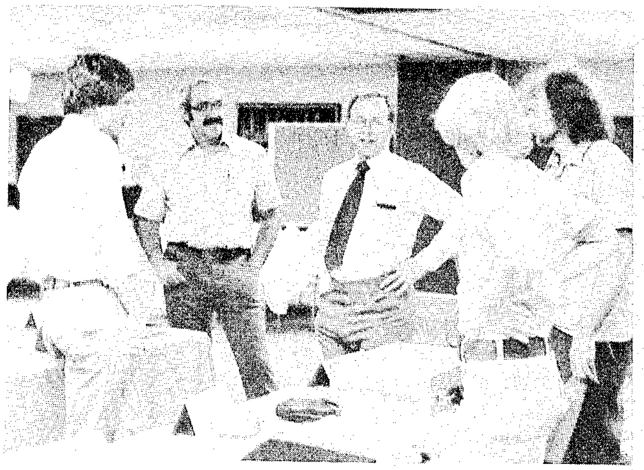
RO-CEL ELECTRONICS

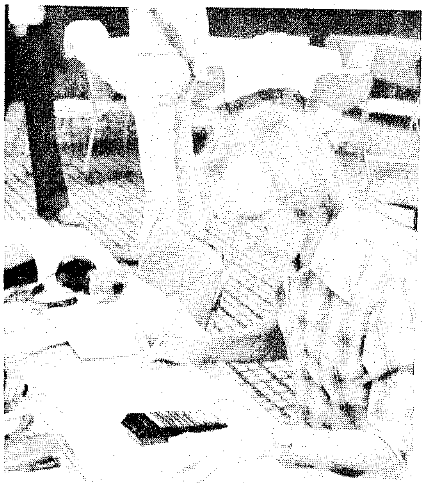


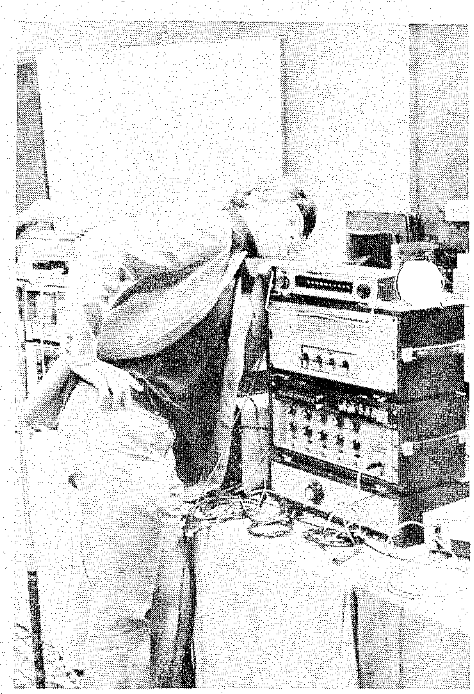
RT 60= 1.5 SEC.	ABSORPTION= 8360 SABINS
SPKR/LISTENER=200 FT.	TALKER/LISTENER=200 FT.
SPKR/MIC= 50 FT.	TALKER/MIC= 2.0 FT.
ROOM VOLUME= 300000 CU. FT.	SURFACE AREA= 30000 SQ. FT.
EAD=10.0 FT.	NO. OPEN MICS= 1

CHICAGO CLASS 1978





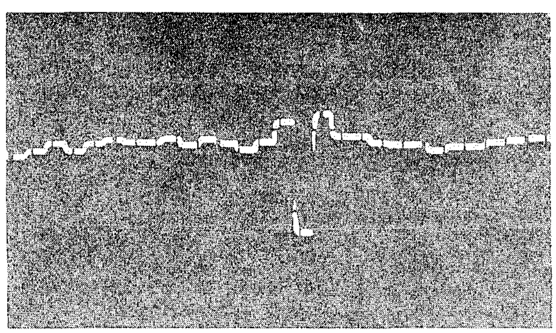




Reprinted from Newsletter
Volume 3, Number 3

MOTHER NATURE'S METHOD

Something to think about the next time someone talks about using boosting filters to tune a room: Dick Heyser has pointed out that no two uncorrelated signals can add in a room to more than 3 dB but they can cancel to any depth. This obvious but often overlooked outcome of adding signals in a real room reminds us that Mother Nature does all of her adjustment of the signal in the room by removing energy, not adding it. A filter bandwidth that does not precisely match the upper envelope of the room's modal structure can cause minimum difficulty if it is attenuating energy but a great deal of difficulty if it is attempting to add power to the system.



The picture on the real time analyzer shows a boost filter on a commercially available 1/3-octave filter set. The 1,000 Hz band is boosted 10 dB and the 1,000 Hz band on the GR real time analyzer is 20 dB in the "cut" position.

DEFINITION OF PHASE

A widely used term in audio these days is "phase". In spite of its widespread useage we find very few engineers who can define phase.

Turning to our trusty IEEE Standard Dictionary of Electrical and Electronics Terms we find:

The phase (of a periodic f(t) for a particular value of t)

The fractional part $\frac{f}{p}$ *of the period P through which t has advanced relative to an arbitrary origin.*

A further note states:

The origin is usually taken at the last previous passage through zero from the negative to the positive direction

$$P = \frac{1}{f} \qquad phase = \frac{t}{p}$$

$$phase\ angle = 360 \times phase$$

Example

A 1,000 Hz sine wave emitted by two sources where the difference in distance between the sources and the measuring point is ¼ of the period

$$P = \frac{1}{1000} = .001$$

$$\text{phase} = \frac{\frac{1}{4}(.001)}{.001} = .25$$

$$\text{phase angle} = 360^\circ(.25) = 90^\circ$$

Additional Comments

Phase is frequency dependent. Polarity *is not* frequency dependent. Phase may be expressed as

$$\text{phase angle in radians} = \text{phase} \times (2\pi \text{ radians})$$

USING THE AVAILABLE POWER CONCEPT

The IEEE Dictionary of Standard Electrical and Electronics Terms defines available power (potential power) as: "Available power (at a port). The maximum power which can be transferred from the port to a load. Note: At a specified frequency, maximum power transfer will take place when the impedance of the load is the *conjugate* of that of the source."

And, under "Impedance, Conjugate -- for an impedance associated with an electric network, the complex *conjugate* is an impedance with the same resistive component and a reactance component the negative of the original."

These definitions are of fundamental importance when calculating "available power" in dBm from open circuit voltage figures. As we have pointed out elsewhere an open circuit voltage has no power *because no current is flowing.* Thus the first step is to match the source's impedance. This results in a voltage divider which places one-half the voltage across the load impedance and one-half across the source impedance. One-half voltage is -6dB.

$$\text{Available power in dBm} = 10 \log\left(\frac{(E_O)^2}{(.001Z_S)}\right) - \text{dB}$$

Where E_O is the open circuit voltage

Z_S is the source impedance

A particularly useful form of the available power concept is the EIA microphone rating which takes the open circuit voltage of a microphone receiving an acoustic input of 94 dB-SPL and then converts it to the available power in dBm for an acoustic input of 0 dB-SPL. This means that adding whatever dB-SPL the performer produces to the EIA rating results in the available power rating in dBm that will appear at the input of the mixing console albeit as a voltage. In other words if, for example, -30 dBm appears at the input and the amplifier has been *correctly* rated for transducer gain -

Transducer gain "the ratio of the *power* that the transducer delivers to the specified load under specified operating conditions to the *available power* of the specified source." Remember, there is no such thing as voltage gain. Then, if the amplifier (transducer) has a 50 dB gain rating, +20 dBm will appear at its output across its specified load.

The EIA Sensitivity Rating

The EIA loudspeaker sensitivity rating provides the same conveniences at the output end of the system. The EIA rating is the dB-SPL at 30 feet that an input of 0 dBm (.001 watt) to the loudspeaker would generate. By having amplifier ratings in dBm instead of in watts (dBm is just another way of saying watts) merely adding the amplifier's output rating in dBm to the EIA sensitivity rating in dB-SPL results in the peak acoustic level possible at 30' from that amplifier and loudspeaker.

For example, if the EIA rating for the loudspeaker is 52 dB-SPL and a 100 watt (+50 dBm) amplifier is used, then the maximum acoustic level possible at 30' becomes 52 + 50 = 102 dB-SPL.

The coherent beauty of the dBm system and its orderly relationship with dB-SPL, dB-PWL, etc., reveals itself to all those engineers who take the time to go past the minor surface difficulties.

SPLIT SPEAKERS GENERATE COMB FILTERS

<u>How Loudspeakers Divided on Two sides of a Stage Generate "Comb Filters"</u>

Let's imagine two loudspeakers separated by some given distance, D_X. Then let's further imagine that a listener sits in front of one loudspeaker at the same distance, D_X, but at an angle to the second loudspeaker. See Figure # 1.

This makes the distance from the listener to the second loudspeaker equal to

$$\sqrt{D_X{}^2 + D_X{}^2}$$

If at this point we were to make $D_X = 20'$, then

$$\sqrt{D_X{}^2 + D_X{}^2}$$

becomes 18.28 feet or a path difference of 8.28'.

A wavelength of 8.28 feet is a frequency of

$$\frac{1130}{8.28} = 136.47 \text{ Hz}$$

Now, if there are 136.47 cycles in 1130 feet, we can then find the number of cycles in 20 feet

$$\frac{136.47}{1130} = \frac{x \text{ cycles}}{20'} = 2.42$$

and in 28.28 feet

$$\frac{136.47}{1130} = \frac{x \text{ cycles}}{28.28'} = 3.42$$

Or, in other words, the signals are one cycle or one wavelength apart.

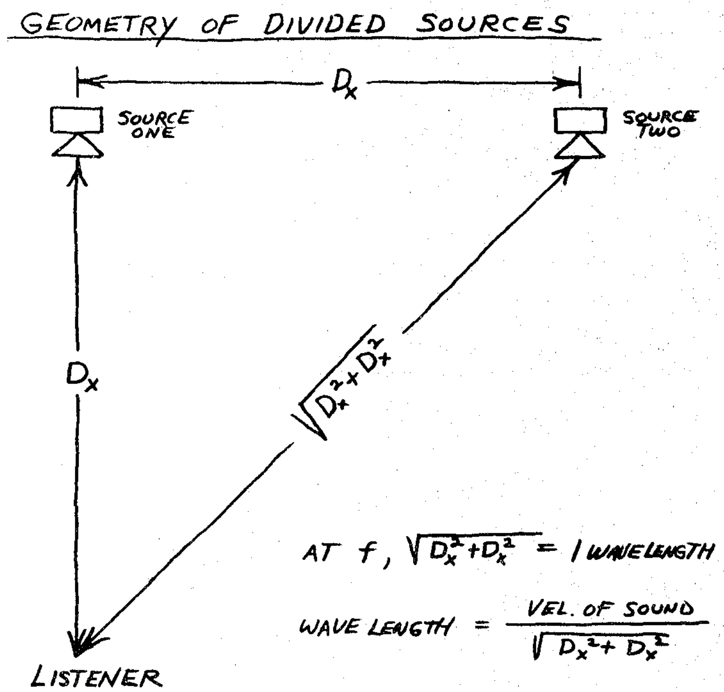


FIGURE # 1

Looking at Figure # 2 it is visually evident that two sources in phase at f but one wavelength apart will be out of phase at $\frac{f}{2}$ and 1.5f, and in phase at f, 2f, etc.

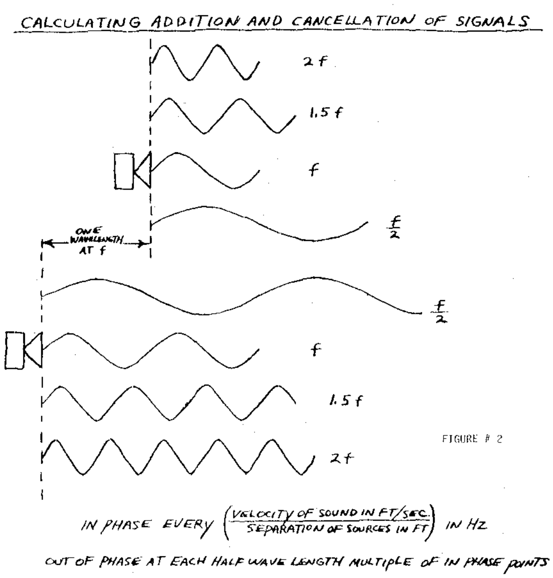


FIGURE # 2

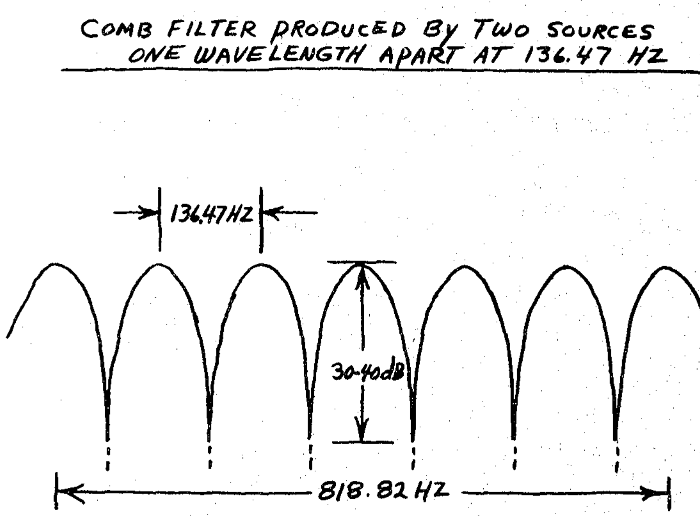


FIGURE # 3

If we were to use an oscillator and sweep from some arbitrary starting frequency for say roughly 1,000 Hz, Figure # 3 shows the kind of amplitude response that would result if only the direct sound were measured. The bandwidths of these amplitude response anomolies are, as has been shown here, directly related to the path length between the two sources.

continued next page

SPLIT SPEAKERS.... continued

$$B.W. = \frac{V}{D}12$$

Where B.W. is the bandwidth
D is the path length distance in inches
V is the velocity of sound in feet/sec

$$B.W. = \frac{V}{D} \text{ when D is in feet}$$

Thus, measuring the bandwidth of an anomoly allows the distance to the actual or virtual second source to be easily and accurately calculated (remembering that in the case of virtual sources, reflections, the physical distance to the reflector may be one-half the acoustic distance when the sound first passes the microphone, travels to the reflector, and then returns to the microphone).

$$D = \frac{V}{B.W.}12$$

Summary

When it is recognized that what we call "room modes" are the complex combination of literally hundreds of comb filters generated by the direct sound and reflections as well as reflections with reflections, then the utility of our 1/3-octave real time analyzers for viewing total sound and our TDS analyzers for viewing direct sound and early reflections becomes apparent and exciting.

Since we can't get rid of anomolies (an anechoic room still has them the minute you move off the center line between two sources) we can now begin the practical and scientific study of how dense they should be and what bandwidths they ideally should be adjusted to. That this can be done "in situ" means a new era in practical acoustics.

SEMI-DISTRIBUTED, SEMI-HIGH LEVEL SYSTEMS

Bob Davis at Altec has put together several quotes from "the industry" on the subject of split speakers:

Burris-Meyer, Harold: Mallory, Vincent; "Sound in the Theatre," Radio Magazine Inc., Mineola, NY 1959, page 19

"Caution: The unforgiveable, and still too common sin is to mount loudspeakers on either side of the proscenium pointed directly at the audience, for simple reinforcement. This system makes the audience, everywhere except on the center line, hear both loudspeakers one after the other, with the result that all illusion of reality is destroyed and in some locations speech becomes unintelligibile and music becomes noise."

Newman, Robert B., Cavanaugh, William J., "Sound Systems", Architectural Record, December 1961

"In good practice today, one rarely locates loudspeakers at the two sides of the proscenium opening, nor does one have a crossfire from loudspeakers distributed down the two sides of the room. These semi-distributed, semi-high level systems never work well, and the hearing conditions in a space can usually be improved by actually shutting them off!"

Davis, Don B., "Pertinent Sound System Practices" Audio Visual Journal of Arizona, 1968

"Never distribute loudspeakers down the sides of walls as this creates artificial echoes over acoustic paths that are excessive in length, and never split speakers on either side of the talker."

Klepper, David L., "Sound Systems in Reverberant Rooms for Worship", Journal of the Audio Engineering Society, August 1970, Page 400.

"Mention should be made of two types of systems that are known to be unsatisfactory for practically any speech reinforcement application: 1) Loudspeakers along the side walls aimed at opposite side walls, and 2) split central systems with the loudspeakers spaced far apart and used simultaneously. These systems either inter-ject too much sound energy into the reverberant space or produce an artificial echo effect. We do not consider them part of our vocabulary or repertory, except for such special purposes as electronic reverberation or sound effects. They are not for speech reinforcement."

Sprinkle, Melvin C.,"Designing Sound Systems," Consulting Engineer, November through December 1970 & January 1971.

"Several types of installations invariably produce unsatisfactory results:...Auditoriums with two loudspeakers, one on either side of the proscenium. This type of installation will always produce somewhere in the auditorium a region extending from front to back in which there is confusion of sound and poor coverage (more cancellation effects). Furthermore, such an installation provides absolutely no illusion of source."

DICK HEYSER EAR OPENERS

Listening to the cassette tapes of the April Graduate meeting we came across the following "ear opener" from Dick Heyser. He is explaining the inability of the most accurate oscilloscope (time domain) or real time analyzer (frequency domain) to tell its operator whether or not a very complex signal is speech, music, or gibberish, and yet a $2 Japanese 4-inch loudspeaker will immediately let the operator's ears and brain detect which.

What parameters from the signal does our brain use to do this? Our analyzers received exactly the same signal but couldn't tell. Where is it hidden in the signal we measure - this difference between speech, music or gibberish?

ARTICLES OF INTEREST

Paul W. Klipsch has published an article entitled, *A Note on Loudspeaker Impedance and its Effect on Amplifier Distortion*, in the July/August 1978 Vol 26, Number 7/8 Journal of the Audio Engineering Society.

Klipsch observes that an impedance mismatched loudspeaker used as a load lowers transient intermodulation distortion (TIM) in a power amplifier. The point he brings up again illustrates the need to consider new techniques of measurement as our present ones do not correlate sufficiently to what we easily hear with our ears. I certainly agree that the proper load on an amplifier's output for meaningful tests should be the loudspeaker it is intended to drive.

We believe that the day is coming in which amplifiers separate from the loudspeaker will be considered obsolete.

*****

The same issue of the Journal of the AES contains another in the series of fundamental historic papers in audio. This time it is the incomparable classic, *Auditory Perspective - Loudspeakers and Microphones*, by E. C. Wente and A.L. Thuras.

Originally published in 1934,few papers can lay claim to as wide an influence as this work can. It is still the starting point for any serious researcher in the field and the few variations that forty years of engineering in this area could suggest as improvements are in minor details only.

Carolyn and I have known some of the men who helped design this demonstration as well as men who were part of the audience. We are told that they still haven't heard better sound even today. We believe them.

*****

There is a not-so-subtle trend in the direction of *appointed* bureaucrats enacting regulations that could never be passed as laws where *elected* officials would have to vote on them.

OSHA has become a dirty four letter word under this philosophy. Currently everything from firearm controls - where ATFB bureaucrats' harassment of legitimate gun collectors at gun shows, false statistics in press releases--to the incarceration of newspaper men and the violent searches of newspaper offices by and at police department discretion when they feel a reporter's source of information is concealed, is part of the frightening trend.

These attacks on basic freedoms, whether you happen to agree or not in a specific case with the intent, are a clear cut danger to us all.

The editorial reproduced here was sent to us by SAM ADAMS of Columbus, GA. Your congressman and senator needs to know that you object to such totalitarian behavior - and we assume you do.

OSHA Spanked Again

by John F. McManus

Belmont, Massachusetts — In the decades prior to the birth of our nation, one of the most hated denials of fundamental rights were King George III's Writs of Assistance. Increasing taxation led to smuggling, and the subsequent avoidance of taxation led to the Writs, which gave bureaucrats and soldiers power to search anyone's home or business.

Our Founding Fathers hated the Writs of Assistance and their well-placed sentiment eventually showed up as Amendment IV of the Bill of Rights. Everyone should read and digest the Fourth Amendment, for it is under attack.

The right of the people to be secure in their persons, houses, papers, and effects, against unreasonable searches and seizures, shall not be violated, and no warrants shall issue but upon probable cause, supported by oath or affirmation, and particularly describing the place to be searched, and the persons or things to be seized.

Bill Barlow's Case

Bill Barlow had not only read the Fourth Amendment, he had posted it on the wall at his plumbing and heating business in Pocatello, Idaho. When a compliance officer from the federal Occupational Safety and Health Administration (OSHA) arrived to conduct an inspection, Mr. Barlow refused him entry because he had no warrant. And when the OSHAcrat came back with one, entry was again refused because the warrant did not specify probable cause.

Then, Bill Barlow sued the federal government. A three-judge federal panel declared the search provision of OSHA "unconstitutional," but the government appealed all the way to the Supreme Court. On May 23, 1978, the Supreme Court ruled that warrantless searches were indeed unconstitutional, but that demonstration of "probable cause in the criminal law sense is not required" by OSHA. OSHA gleefully interpreted the decision as a directive that warrants must be obtained, but merely for the asking.

Weyerhauser Decision

Even before the final Barlow decision, America's businessmen had begun to use the Fourth Amendment to defend themselves from OSHA's fishing expeditions. On June 30, 1977, OSHA was denied admittance to a Weyerhauser box manufacturing plant in Wisconsin. When the OSHAcrat returned a week later with a warrant, but without "probable cause," he was admitted "under protest." Then, Weyerhauser went to court.

On July 3, 1978, U.S. District Judge Myron L. Gordon ruled that it is not enough that OSHA be satisfied that probable cause for an inspection exists. He interpreted the Barlow decision to mean that before any warrant can properly be issued, the judge to whom OSHA applies must also be satisfied that "probable cause" exists.

Judge Gordon then voided the warrant used in the Weyerhauser inspection, quashed all citations and penalties imposed as a result of the use of that warrant, and ordered OSHA to return to Weyerhauser whatever evidence it had accumulated through an illegal search.

What This Means

Because of the Barlow and Weyerhauser decisions, businessmen can *and should* deny entry to OSHA unless there is a search warrant. Further, they should insist that any warrant be issued upon demonstration of probable cause before the judge, not simply at the request of an OSHA bureaucrat.

OSHA has had some large holes punched in it. But it should be abolished. If enough Americans become familiar with the Fourth Amendment, and why it was written, the day will soon come when OSHA will share a grave it so richly deserves — right alongside King George's tyrannical Writs of Assistance.

BOOKS OF INTEREST

WALT DISNEY - AN AMERICAN ORIGINAL, by Bob Thomas, published by Simon & Schuster. $9.95.
A recent view of this book in the SMPTE Journal by a man ensconced in a non-profit making organization quotes various *former* employees of Disney and concludes "Among the facts Thomas overlooks, or alludes to only veiledly, are Disney's autocratic nature, his pitiless perfectionism, the cruelty that pervades his early works." This same reviewer complains about the salaries Disney paid "inkers" in 1938. Complaining about 1938 wages should only be done by someone who knows what it meant to make any kind of *honest* money in 1938.

Perhaps neither the book written by Thomas or the reviewer quoted above tell the whole story but the book comes close.

Very few startlingly creative people are easy to live with. They exhibit a singlemindedness of purpose, a pursuit of perfection, and one other trait totally foreign to the critically minded, that of possessing a dream. *They* know where they are headed even if the others not only don't know but in most cases don't care. Someone has said, "Be careful of your goals in life - you might get them".

Disney's goals were set so high that dedicated followers are still reaping exceptional profits pursuing "left over" ideas from before Disney's passing twelve years ago.

Disney actually *did more* than ordinary men dream about. No wonder the pitiless critic reveals jealousy, pettiness, and total lack of understanding when reviewing a book favorable to a giant like Disney.

We found Bob Thomas' book reflected a spirit true to what we can regularly view of Disney's work. The greatest tribute to Disney other than his work is that he offended the right people.

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The reason for giving you this is so that in the future you may think of someone else, other than yourself. Besides, I don't like domineering, egotistical or simple minded drivers and you probably fit into one of these categories.

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