SYN-AUD-CON
SYNERGETIC AUDIO CONCEPTS

newsletter

VOLUME 6, NUMBER 2
January, 1979

P.O. BOX 1134, TUSTIN, CALIFORNIA 92680

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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A SPECIAL OPPORTUNITY FOR 20 TDS LEADERS

Tentative plans are underway to have a very special Workshop in May, the week before the West Coast AES Convention, in Hollywood at the Wally Heider-Filmways Studio 4 for twenty (20) TDS licensees.

We apologize for the limit of twenty but it is a firm limit as the intention is that each of these 20 will have "hands on" training in TDS, FFT, LEDE, TA^{tm}, PRP^{tm}, and PZM^{tm} in a state-of-the-art new LEDE control room featuring UREI 813 time aligned monitors, a new Nevecam console, and digital tape recording.

The instructors are to be:

DICK HEYSER on TDS-FFT

ED LONG on TIME ALIGNMENT (shared with Ron Wickersham)

RON WICKERSHAM on PRP AND PZM (shared with Ed Long)

DON DAVIS on LEDE

CHIPS DAVIS on LEDE AND PZM (assisted by Ken Wahrenbrock)

If you are to be one of the twenty participants you must be licensed to practice TDS, prepared to pay $750. for the three days, and willing and able to start from scratch and go from there to way beyond the current state of the art in sound technology.

The $750 fee covers your class materials, morning and afternoon coffee breaks, lunch, and dinner each of the three days. A special certificate will be awarded at the completion of the Workshop. All transportation, housing, etc. is the responsibility of the attendee but we will set aside hotel space for those requesting it.

The majority of the three days will be spent by *you* actually operating all the equipment, by *you* analyzing the data taken, and by *you* interfacing for significant periods of time with your instructors. That's why we *must* limit this opportunity to only 20. (We will have several full sets of TDS and FFT equipment.)

We don't know at this time if it will be possible to do such a Workshop more than once. It's extremely difficult to get premier studio space for three days or to gather together for three dedicated days a staff of this caliber with prepared material to share exclusively with you.

Because of the extremely complex scheduling and coordination of efforts, we must know by March 15, 1979 if you wish to be one of the twenty. The first twenty are it. After that we will put ten more on a standby basis in case of unexpected cancellation. Half of the fee, $375, is due upon your acceptance. Don't send money until you are notified that your application was received in time. The remainder of the fee is due two weeks prior to the meeting. Applicants will be accepted in the order received. If more than one person registers per TDS licensee, indicate who is the first choice to attend as the second person will be put on a wait-list pending available space.

We will send the successful applicants information on hotels, transportation, etc., relative to the meeting area. We will work from 9:00 a.m. to 9:00 p.m. each of the first two days and until 6:00 p.m. the third day, followed by a graduation dinner and awarding of "the 20" certificates.

We believe that "the twenty" will leave uniquely equipped to lead the audio industry into a whole new era of sound technology.

If you are not currently a TDS licensee, your application for this Workshop will be considered *if* you attach an application for a TDS license as well (see details below).

We will not finalize these plans until we have received the first twenty applications, by letter or phone. We will provide you official registration material as soon as we have 20 acceptances; therefore, this opportunity is a tentative offer only, though your instructors and Studio 4 at Wally Heider-Filmways are available May 9-11, 1979.

Those applying will be notified directly after March 15 (or before if 20 are registered) as to the status of this TDS Workshop.

TDS LICENSEES

At this time there are 62 TDS licensees. In our October 1978 Newsletter we listed 33. If you would like to license to practice TDS, make out a check for $100 to California Institute Research Foundation. Make out a check for $5 to Syn-Aud-Con for the special package of data we have prepared for the licensees. Send both checks to Syn-Aud-Con and we will process your TDS license with the California Institute Research Foundation.

SYN-AUD-CON EDITORIAL

January 1979 marks the beginning of Syn-Aud-Con's seventh year of service to the audio industry. How exciting it is to find ourselves in the middle of Audio's second "golden era" - the first was making the motion picture "talk".

What makes the present era "golden"? You are part of the largest group of professionals ever to dedicate their talents to our industry. You, the hard working, willing to study, tremendously innovative, and generously sharing Syn-Aud-Con graduate are making your era "golden".

Think of the role Syn-Aud-Con and Syn-Aud-Con graduates have played in products like the Crown and Ivie real time analzyers, in new ideas like KEN WAHRENBROCK's PZM adaptation of Ed Long and RON WICKERSHAM's PRP^{tm} techniques, and think further: last year TDS had one licensee and now approaching 75 licensees. Talk with any manufacturer and you will find that Syn-Aud-Con graduates are the most cussed and discussed group of individuals in the industry. Your knowledge of what is relevant is causing products, specifications, and sales approaches to change for the better.

Most exciting of all is the realization that the first six years is but the preface to a whole new way of doing audio business. In professional audio, product is important, but not nearly so important as how to apply it, install it, service it, and adjust it. Syn-Aud-Con graduates lead the world today in *their knowledge* of audio. Those who really wish to know how to scientifically design commercial sound, or professional recording systems must come to you or suffer second place. We're proud of all of you and will work even harder during the next six years sharing the excitement you generate.

SYNERGETIC AUDIO CONCEPTS

GENRAD 1921 REAL TIME ANALYZER FOR SALE

Syn-Aud-Con's large General Radio 1921 Real Time Analyzer is for sale.

This is the large real time analyzer we use for microphone testing, etc. It consists of three major components:

1. The 1925 Multifilter - 30 parallel 1/3-octave filters fully adjustable over a 40 dB range

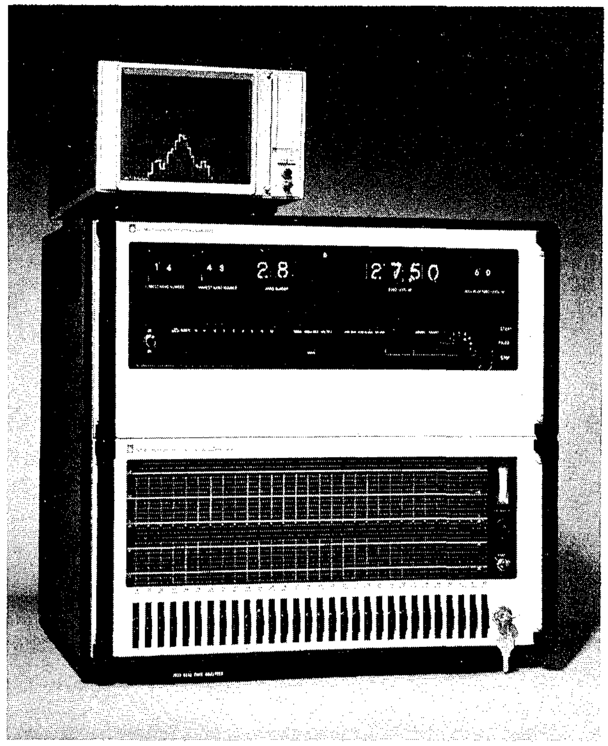
2. The 1926 Multichannel RMS Detector, 1 Hz to 100K Hz. 70 dB dynamic range, 60 dB display, digital detection. True RMS answers, fully programmable

3. The 1921-P2 Storage Display Unit. This is the Tektronix special display capable of showing literally hundreds of displays on its screen, calibrated from 20 to 20,000 Hz over a 60 dB dynamic range

The three units are in exceptional condition.

Price: $6,000.

We will have the local GenRad office pack for shipment if the purchaser is outside comfortable driving distance of Tustin.



SYN-AUD-CON GRADS HELP IN CLASSES

During our Fall 1978 tour volunteer graduates helped us in our classes. This is hard work that requires dedication, drive, and a desire to participate in the inner activities of Syn-Aud-Con. It takes almost 8 hours to unload and about 5 hours to disassemble and pack after a class ends. There are many tasks in helping newcomers in the classes, assisting in demonstrations, working with groups in the evening sessions, etc.

At St. Louis (special Rauland class) we had a southern accent attached to a real dynamo named SCOTT LAUCK who flew up from Lafayette, LA. In our Chicago class, TOM HAYES, a two time grad from Champaign, IL provided the muscle and mind needed to meet the needs of an exceptionally interesting class. GLEN BALLOU of Southington, CT is in a special category - more like a close relative than friend - helped us in our New York class this year.

The remainder of the classes in the East (DC, Atlanta and the two classes in Orlando - one special for Rauland)-- we were assisted by a young couple, GINA and FARREL Becker. They bought a trailer and traveled from class to class with us. Since we travel with Punch and Judy, they brought along Nigel who was such a gentleman that he violated a habit of long standing and refrained from chasing cats while Punch was present.

FARREL & GINA

We were delighted to have Farrel and Gina, two very talented, considerate and genuinely creative young people with us.

One last special Rauland class was held in Anaheim in December and KEN WAHRENBROCK of PZM fame worked with us and helped us "polish" the new "Sound System Design Worksheet" that Farrel and Gina helped us re-work for the Rauland class in Orlando. (We hope to publish the new version in April)

Carolyn and I are immensely grateful to these graduates for their help, their talents, and their interest in Syn-Aud-Con.

LEDE CONTROL ROOM AT SUMMIT UNIVERSITY

The quotes below are from a letter received recently from Harry Spielberg of Summit University in Malibu, CA. Their construction of an LEDE control room was undertaken from seeing Tech Topic Vol 5 # 7 and a short personal discussion during the course of consultation on a church auditorium system. We are pleased to hear proof that good basic ideas result in better listening conditions.

> You may recall that you were here over the summer on our new campus "Camelot" in Malibu Canyon to measure and counsel us on a system for our chapel. We have followed your advice and the results are very pleasing.
>
> Since then we have built a true LEDE control room as our recording and distribution center for the entire campus. We have followed your design parameters and we are extremely happy with it. The sound is very clean, natural and intelligible. There is no forced or artificial sound or anomalies. We find it very easy to listen in for extended periods as the naturalness reduces listening fatigue. Because of the great success of our control room we have already begun to construct our mastering room in the LEDE design. This design is truly a new era for natural acoustical control facilities, and its beauty is also in its simplicity. It is not as fancy as a full-budget commercial studio but the design has been closely followed and the effects are beautifully obvious.

UPDATED FILTER FORMULA

TED UZZLE sent in a useful re-write of the equation for finding the capacitor value associated with protecting high frequency units in biamped sound systems shown on page 124, eq 7-24 of SSE.

$$c = \frac{10^6}{\pi f(Z_S+Z_L)}$$

Where c is the capacitor value in microfarads (ufd)
f is the crossover frequency of the biamp crossover
Z_S is the internal impedance back into the amplifier
Z_L is the loudspeaker impedance

This equation places the -3dB point of the capacitor one octave below the biamp network's for both matched *and* unmatched cases.

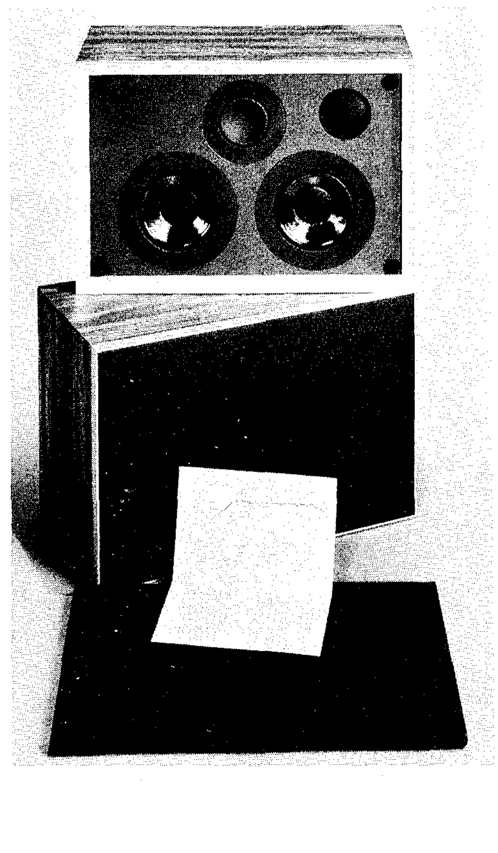
Example

Loudspeaker is 8Ω. Amplifier's true output Z is .1Ω. What value capacitor do I need if the biamp network crosses over at 800 Hz?

$$\frac{10^6}{\pi(800)(8+.1)} = 49.12 \text{ ufd}$$

"MIX DOWN MONITORS"

Ed Long stikes again! Ed is involved in still another interesting facet of loudspeaker design. He has developed a near field monitor (NFM)tm for use as Mix Down Monitors. Ed calls these units the Model MDM-4. They come with extensive calibration charts. We have not, as yet, had the opportunity to test these units but feel that the concept is a sound one worthy of further investigation.



PERFORMANCE SPECIFICATIONS:	DESIGN SPECIFICATIONS:
FREQUENCY RESPONSE ±3 dB 70 Hz to 17 kHz (4 π steradians, free field) ±5 dB 60 Hz to 20 kHz (4 π steradians, free field) POWER REQUIREMENTS 10 watts for 97 dB per 1000 cubic feet or room volume 1 watt for 89 dB SPL @ 1 meter (free field) POWER HANDLING 40 watts continuous 100 watts instantaneous below 1 kHz 15 watts continuous 40 watts instantaneous above 1 kHz DEMAGNETIZATION LEVEL 315 watt low frequency pulse will result in a permanent 1 dB output reduction in the piston band of the bass drivers SENSITIVITY 80 dB SPL/volt/meter DISTORTION Less than 5% THD or I.M., 60 Hz to 20 kHz Typically less than 1% 100 Hz to 20 kHz 94 dB SPL at 1 meter IMPEDANCE 8 ohms nominal 5 ohms minimum	SYSTEM TYPE Dual woofter, 2-way system with Velocity Control High Pass Filter DRIVERS Two 16 cm (6½") Low Frequency One 9 cm (3½") Mid-high Frequency CROSSOVER Equalizer Filter type at 1500 Hz ENCLOSURE VOLUME 27 Liters (95 cubic feet) ENCLOSURE DIMENSIONS 48 cm x 33 cm x 24 8 cm 19" x 13" x 9¾" ENCLOSURE FINISH Rosewood Laminate with Aluminum Trim GRILLE Brown cloth SYSTEM WEIGHT 10.4 kgm net, 11 3 kgm shipping 23 lb net 25 lb shipping

CALIBRATION STANDARD INSTRUMENTS
PO BOX 2727 OAKLAND CALIFORNIA 94602

J W DAVIS CO. TO PRODUCE HEYSER INVENTION

The J. W. Davis Company in Dallas, Texas is preparing to announce production of Dick Heyser's SBA system. SBA stands for "Signal Biasing Amplification". SBA is not a device but a total system's concept, a concept of *distributed amplification.*

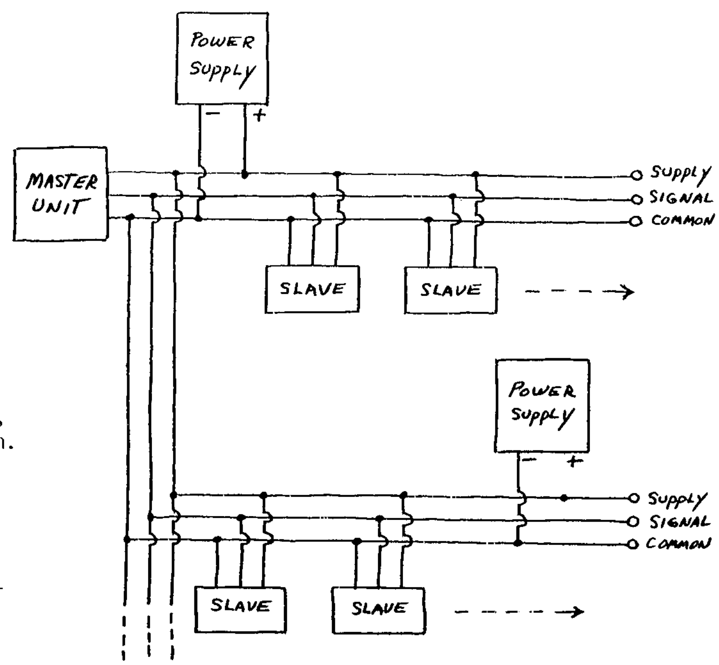
In this system there are no high powered amplifiers, amplifier racks, amplifier rooms, special air conditioning needs, etc. The Master unit in this system neither has nor requires a power amplifier. The conversion of electrical energy to audio power is accomplished at the "slave" amplifiers at the loudspeaker. However, this *is not just a power amplifier at the loudspeaker type approach.*

The manner in which this system operates is that the master unit provides a special kind of signal which contains the audio program *plus* a precisely generated bias voltage. The bias is determined directly from the program signal and varies continuously with it. This action gives the system its name: Signal Biased Amplification.

To quote from Dick Heyser's description of this concept: (excerpts from a Dick Heyser report on SBA)

> The bias voltage is the precise dc level required by each slave in order to operate in a linear class A manner, yet consume the least amount of power supply current. The amount of power supply current is always the exact amount needed to provide the instantaneous signal which each speaker is called upon to handle. Power consumption varies directly with signal. When there is no music signal, there need be no power supply drain.
>
> Each slave/speaker can have its individual level controlled by a simple high impedance potentiometer. As the gain is reduced, the instantaneous bias is reduced accordingly. Less power is consumed as the sound output of the slave is reduced. If the level is turned all the way down, the system signal power drain is zero. If a system had ten thousand speakers, and each speaker were turned to no output, there would be no signal power drain even though the entire net were energized by the master. If one speaker were then turned up, the power drain of the entire system would only be the amount required by that one speaker and would appear only at the place where that speaker was located. As all the other speakers were turned up, the system power drain would rise accordingly to meet the instantaneous need.
>
> The raw power of an SBA system derives from low voltage dc power supplies distributed throughout the net. Three wires are needed to carry the common signal, and dc supply. The minimum power supply voltage is determined from the maximum peak to peak voltage which is needed by the loudspeaker. Where a few watts per speaker is the maximum anticipated demand, the line voltage can be held to twenty volts or less, thus falling below the 25 or 70 volt limits of many local codes. This can reduce the installation cost.
>
> Each master can control many hundreds of slaves. In turn, each slave can act as a submaster and control a like number of subslaves. This process can continue indefinitely.
>
> There is no signal transformer used in the SBA net and frequency response extends from dc to many tens of kilohertz. (ED: underlining mine) Topology of the interconnecting wires in an SBA net can be configured such that the resistance of the wire contributes to a distributed amplifier mode. Thus in a long chain of slaves, the signal voltage at the slave farthest from the master can be equal to or larger than the signal voltage coming out of the master itself.
>
> Executive control of the net comes from the master amplifier. But the master amplifier can be placed in any convenient location in an SBA net. It is possible for two or more master amplifiers, widely distributed in the net, to share executive control. Since the master does not supply signal power, and indeed can take whatever supply current it needs from the net itself, the master can be a hand carried unit no larger than a pack of cigarettes.
>
> These and many other unique features of an SBA system, provide an entirely new concept in sound system design. The SBA systems concept has been in existence for almost twenty years. It is not some new blue-sky idea. However, the first, and most important, fact to bear in mind is that an SBA system provides good clean sound. Think of this in terms of the high fidelity sound quality which it provides. After gaining some feel for the quality of service it offers, then we can begin discussing some of the things an SBA does which no other system can do.

J.W. DAVIS - HEYSER SBA SYSTEMS CONCEPT



Syn-Aud-Con has seen and heard the SBA system. We believe that it *obsoletes* the current approach to low level distributed system for paging, background music, etc. If you have any occasion to design, install, or service such systems, we urge you to become one of the pioneer firms lucky enough to get in on the ground floor, by writing or calling:

R. H. Chapman
Manager-Electronics
J. W. Davis Co.
9212 Denton Dr.
Dallas, TX 75235
214/352-8405

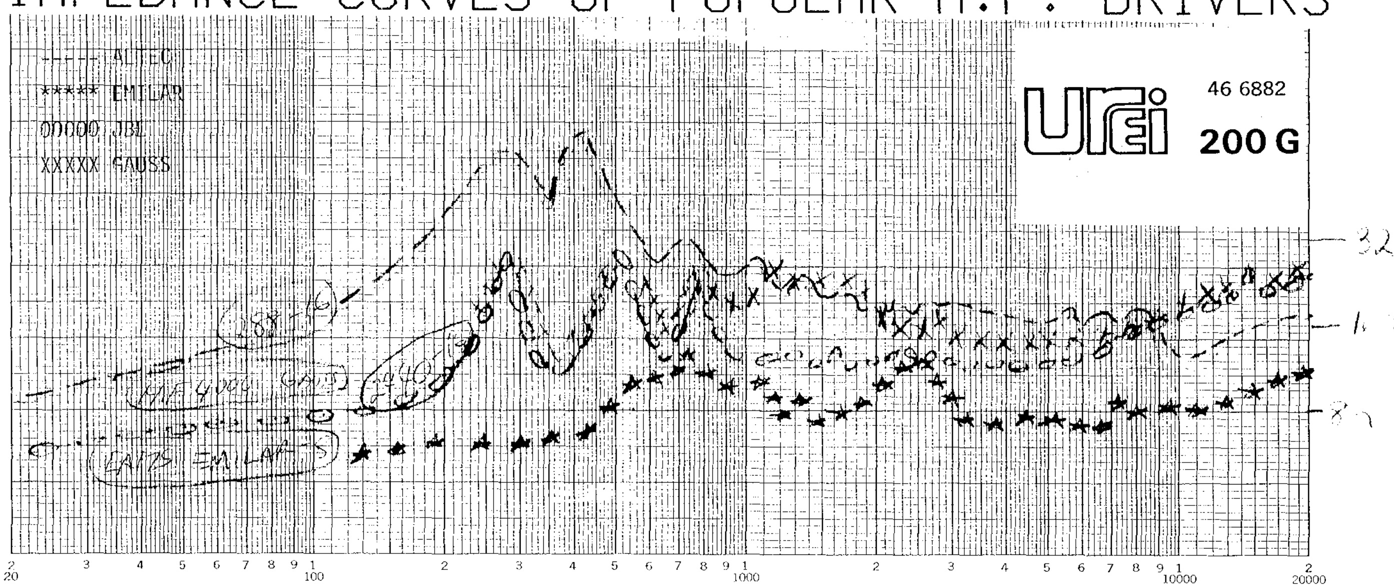
"CULTS"

Erich Fromm divided the human race into two orientations called necrophile and biophile. A necrophile is death centered - the aunt who loves to attend funerals, the guy who is fascinated by disasters. The biophile is life centered, not afraid of normal living hazards, willing to be responsible for his own actions and in general enjoys the predominant good in the world.

The necrophiles have come out from under their rocks to attack "cults". You'll soon find that a "cult" is whatever they think they can successfully attack. The type of security they have in mind is found in the prison and the grave. Men left free to do as they wish often will abuse that freedom. In our opinion, it is intellectually insupportable to blame freedom. We distrust anyone who thinks we need the "protection", so called, of a tyrannical law depriving us of a free choice be it good, bad, or indifferent in other's eyes. The real problem is the failure to live up to freedom's promise. One should not be hypnotized into attacking their neighbors' beliefs just because they're different.

There will always be those who think that a new law will correct human fraility but history has recorded their efforts as futile. While persecution of religious and philosophical groups as cults will keep legislators off the streets for awhile, it can have tragic results for the victims they choose to exploit for their own publicity efforts.

IMPEDANCE CURVES OF POPULAR H.F. DRIVERS



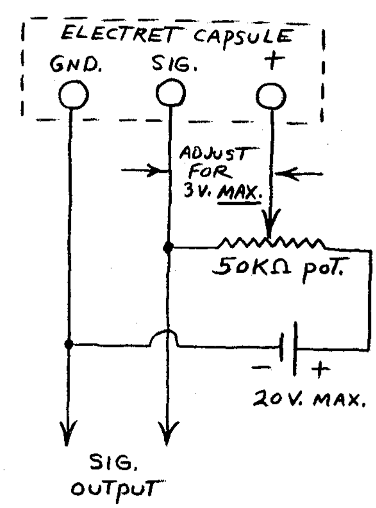
A series of impedance curves of popular H.F. drivers. No wonder Emilar has grown so rapidly.

IMPROVING THE DYNAMIC RANGE OF PZM SYSTEMS

Because of some special internal circuitry associated with the builtin FET in the miniature electret capsules used in these microphones, a simple increase in supply voltage does not result in an increase in their dynamic range.

The diagram shows how to achieve the desired result.

These electrets normally begin to distort at 110 dB-SPL when set to low voltages such as 1.5 volts. If the arrangement shown here is used, the same level of distortion will then begin at approximately 140 dB-SPL (when close miking drums, levels of 130+ are encountered).



LENZ LAW

In 1834 Heinrich Friedrich Lenz (1804-1865) deduced that for a coil moving in a magnetic field: "The induced current will appear in such a direction that it opposes the change that produced it."

A moving loudspeaker is a good example of Lenz's Law in its generation of a "reverse EMF", as a product of its motion at resonance, that appears to the amplifier trying to drive the loudspeaker as an increased impedance.

This "motional" impedance is verified by "blocking" the motion of the cone and seeing the impedance "magnitude" drop to near the ACR value.

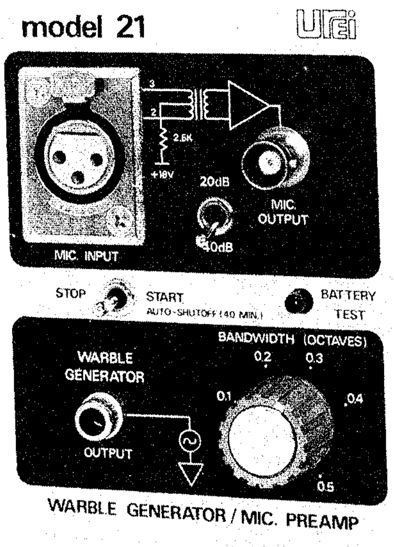
MORE PICTURES FROM CHICAGO & NEW YORK

SYNERGETIC AUDIO CONCEPTS

THE INPUTAMPGRAPHICEQUALIZERFEEDBACKSUPPRESSORBIAMPNOISE-GENERATORCROSSOVERNETWORKDISHWASHINGANDHANDHOLDING COMPONENT

UREI is one of those delightful companies small enough to have extra bright, highly involved engineers able to seek out needs in the market place and clever enough to respond to them with innovative designs that are right the first time.

On the other hand, they're big enough to be sufficiently well financed that they can make a lot of a new product at a very competitive price - but then that's what they're known for, isn't it?



WARBLE GENERATOR / MIC. PREAMP

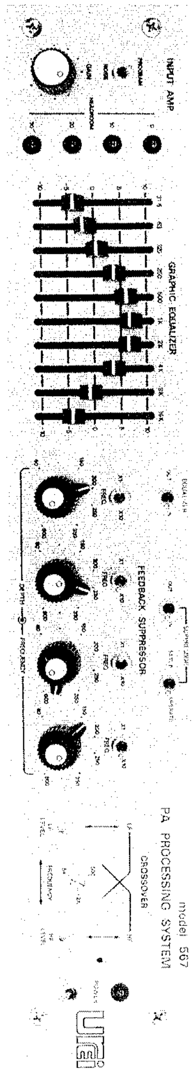
The two latest "Why didn't I think of that?" products are shown here.

I can't describe the number of times I've needed a single channel measurement quality microphone preamp. Hurrah and a 21 gun salute for putting it into one of the most useful components we use, the UREI warble generator.

Note that once its turned on it stays on for 40 minutes and then shuts off. I wish all my battery operated equipment did this.

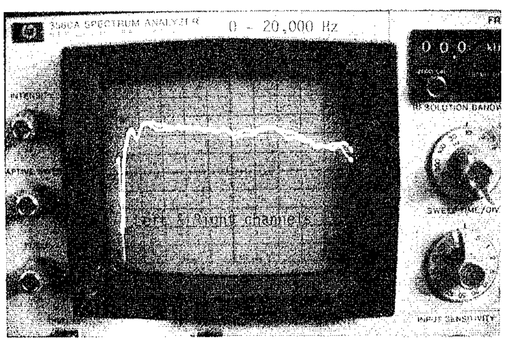
The second idea is a panel that in German would translate back into English as "The inputampgraphicequalizerfeedbacksuppressor-biampnoisegeneratorcrossovernetworkdish-washingandhandholding component. What a marvelous portable system component.

Think about these two products. All the individual parts are already proven in the field. But, no one has put them together until UREI did. That's what makes UREI an interesting company with which to be associated.



LIVING WITH A UREI 813

The UREI 813 monitor loudspeaker for control rooms is being recognized as a vastly superior loudspeaker system. Recording engineers we have encouraged to try these systems report back that they had no idea how fundamental time alignment^{tm} is to the solution of what a monitor *must* do.



We are increasingly involved in control room redesign these days because of LEDE; therefore, we felt it would be wise to undergo "calibration" in terms of the 813 by listening extensively to familiar recordings in a given environment. **BUD MORRIS** of UREI graciously loaned us a pair of 813's for this purpose and we installed them in one end of our living room up in the country (not an interior decorator's dream, but neither is our house). It is apparent that UREI's manifestation of Ed Long's time alignment deserves, and we're sure will receive, a special niche in the history of recording tools.

The UREI-813's are devastating to a poorly made recording, and there are some ghastly examples on today's market. But, pick out an exceptional recording, preferably voice, and you will be quickly transported from the realm of critical evaluation to musical Euphoria.

Later we were at Filmways-Wally Heider large ex-RCA control room which, among other features, has a huge *curved* window to the studio. We heard their new UREI-813s perform. DAVID BRAND, chief engineer at Filmways-Heider Recording, demonstrated convincingly that at any level we could bear that they could reproduce *ultra*-cleanly.

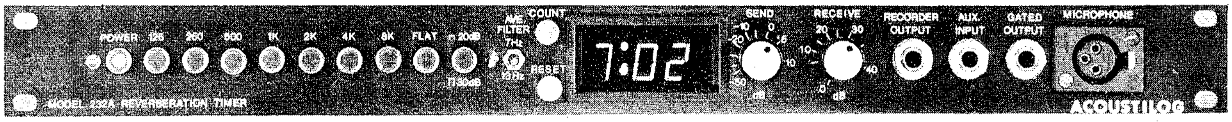
In another of their control rooms they have set up an A-B test switch between a standard 604 with a Mastering Lab crossover and with the UREI time align network. On master tapes the switchover is definitely not subtle; it's dramatic. It's a repeat of Dick Heyser's Catastrophe Theory - once you have heard and identified a distortion, never thereafter are you going to be satisfied to live with it.

CHIPS DAVIS is the latest enthusiastic convert to UREI-813s. He also is the first to have a complete coherent chain from one end to the other: PZM in the studio, TA in the LEDE control room. Chips is in the position of the man "who knows". Until you have a chance to hear the whole chain, you're basically not state-of-the-art in tomorrow's sound.

ATLANTA CLASS

SYNERGETIC AUDIO CONCEPTS

ACOUSTILOG 232A REVERBERATION TIMER



AL FEIERSTEIN is the founder-owner of Acoustilog. He is shown here holding his latest success, the 232A Reverberation Timer.

Rarely have we so enjoyed testing a new version of a product, especially since we were so familar and happy with the previous model. The two LEDs, 4 dB apart, make level setting fun and super accurate. The versatility of being able to press combinations of the filters for 1/1-octave and 1/3-octaves plus the much clearer markings on all the controls, lets you take precision measurements with remarkable speed.

Option 09, in our judgment is an absolute must for any work in a control room environment. The presence of diffusion or more importantly, its absence, shows up clearly on the oscilloscope decay display as a train of discrete reflections.

Al's 232A is outstanding for work in recording studios, control rooms, and other small semi-reverberant environments where slope rates must be both observed and measured accurately. The basic Timer without options sells for $795.

Specifications

Display Range	9.99 seconds
Decay Selector	20 dB (x3) 30 dB (x2) for greatest accuracy
Time Base	Quartz crystal oscillator
Timing Accuracy	±2%, ±3 counts
T_{60} Range Accuracy	± 2 dB
Filter Center Frequencies	63 Hz-250 Hz in octave steps 400 Hz 12.5 KHz in 1/3 octave steps Accuracy ±10%
Filter Bandwidths (-3dB)	One octave each, half octave overall (send and receive)
Noise Source	Built-in pink noise generator, 20 Hz to 20 KHz, ±2 dB
Recorder Output and Gated Output	Variable, 3 volts RMS into 600 ohms or greater
Automatic Level Detection	2 LEDs display 4 dB window
Microphone Input	Electronically balanced, low impedence 15 v Phantom-powered for AKG C451E. Adaptable to line level
Auxiliary Input (for use with external signal source)	10 K ohms unbalanced, 30 dB amplification (variable)
Size	19" (48 cm) wide, 1 3/4" (4.4 cm) high, 7" (17.8 cm) deep
Weight	3 lb (1.4 Kg.)
Power Requirements	100-125 v, 50-60 Hz, 5 watts 220 volts optional

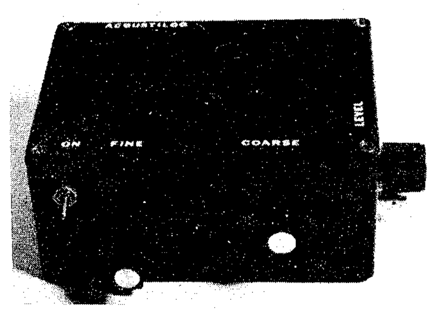
ACOUSTILOG, INC.
19 Mercer Street, New York, N.Y. 10013 (212) 925-1365

ACOUSTILOG "BLACK BOX" FOR TDS WORK

As if the new 232A isn't enough, Al used his spare time to come up with a special "black box" for TDS work that is reliable, stable, battery operated, small, and has a precision ten turns pot that makes TDS tuning easy.

VCO - 1 TIME-DELAY SPECTROMETRY OSCILLATOR

The ACOUSTILOG VCO - 1 is a highly stable voltage-controlled oscillator designed specifically for use in Time Delay Spectrometry. Used in conjunction with an HP 3580A Spectrum Analyzer and a frequency counter, it provides control over the time-delay distance and thus enables acoustic events to be measured. The VCO - 1 is battery-powered and is self-contained in a miniature diecast aluminum case.



Controls =	Power Coarse Tuning Fine Tuning Output Level Control BNC Output Connector
Batteries =	Two Standard 9 volt cells.
Size =	4 3/4" (12 cm.) X 3 3/4" (9.5 cm.) X 2 1/4" (5.5 cm.)
Weight =	20 oz. (0.6 kg.).
Price =	$ 225.00. Shipping Charges = $ 4.00. A copy of your TDS patent license must accompany your check or money order.
Delivery =	2 Weeks.

At $225 the VCO-1 is $470 less expensive than our other choice for the same work, and the $695 unit is not battery operated. PLEASE NOTE: You must be licensed to practice TDS before you can buy Al's VCO-1 Time Delay Spectrometry oscillator. With graduates like Al working on it, TDS can and will soon be as common and easy to use as present day RTAs.

SYNERGETIC AUDIO CONCEPTS

RT_{60}B REVERBERATION TIMER

VIC HALL and his talented engineer, DAVE JOHNSON, at Communications Company in San Diego, have come up with an excellent RT meter to measure reverberation time in churches, arenas, and other large reverberant environments. The RT_{60}B has the ability to obtain accurate reverberation measurements from handclaps or two books being slapped together.

Other features are visual indication of too high, medium, or too low level settings, plus error indication as well as the RT_{60} to two decimal places.

Small as a slightly inflated handheld calculator, it is a good choice for everything except recording studio work where more than 1/1-octave band resolution is required and logarithmic display of the decay on an oscilloscope is a necessity.

Well equipped engineers will find a way to own both Communications Company RT_{60}B and the Acoustilog 232A. Both represent an excellent choice for the special problem areas they were designed to solve.

The RT_{60}B sells to Syn-Aud-Con graduates for $550.00. Write Communications Co. 3490 Noell St., San Deigo, CA 92110. (714) 297-3261

MORE ON S/N-PEUTZ EQUATIONS

ED LETHERT has isolated the S/N portions of the Peutz equations in terms of Figure 4-22 in SSE. We are most grateful for this work as it took a thorough understanding of the underlying parameters and a skillful use of algebraic manipulation to isolate these equations. Contributions of this type are fundamental to the improvement of SSE and Ed's derivations will be included in the next edition with a special footnote acknowledging his excellent work.

$\%Al_{cons}$ Correction (S/N < 25dB)

Increasing S/N to figures greater than 25 dB *does not* improve $\%Al_{cons}$ but in rooms with the $RT_{60} \geqq 1.6$ secs. Any lowering of the S/N below 25 dB can have immediate detrimental results.

To correct the $\%Al_{cons}$ calculation obtained through the use of the Peutz equation (which always assumes a minimum S/N of 25 dB) when the S/N $\leqq$ 25 dB.

$$\%Al_{cons}\ (S/N \geqq 25dB) \times \left(\left(\frac{1}{.09RT_{60}}\right)^{\left(\frac{25-S/N}{35}\right)}\right)$$

Minimum Allowable S/N to Maintain 15% Al_{cons} or Other Desired Values of Al_{cons} When $RT_{60} \leqq 1.6$ secs

This equation provides the mathematical equivalent of Fig. 4-22 on page 71 of SSE.

$$S/N_{min} = \left(\frac{(2-\log \%Al_{cons})35}{(2-(\log 9 \cdot RT_{60})}\right) - 10$$

Example # 1

You have calculated for $15\%Al_{cons}$ in a 2.5 sec room but upon measuring the ambient noise level at installation time you find that you will only have 15 dB S/N rather than the 25 dB you had planned on unless the HVAC contractor is forced to quiet his system. In order to obtain the architect's and owner's attention to the problem you need to calculate how much degradation this excess noise will cause the sound system's proposed performance.

$\%Al_{cons}$ (S/N $\geqq$ 25 dB) = 15% (which is what you planned as a *maximum* compromise)

$$15\% \times \left(\left(\frac{1}{.09(2.5)}\right)^{\left(\frac{25-15}{35}\right)}\right) = 22.97\%$$

This means a substantial number of listeners will be cupping their ears and complaining legitimately that they can't hear *unless* the noise level is lowered (usually by "balancing" the air conditioning system).

Example # 2

Here you are using the $\%Al_{cons}$ equations in a small over-damped hotel meeting room (RT_{60} = .4 secs) which would like to plan on as low a sound level as possible while maintaining intelligibility. What minimum S/N can you plan on in this case?

$$S/N_{min} = \left(\frac{(2-\log 15)\ 35}{2-(\log 9(.4))}\right) - 10 = 9.97\ dB$$

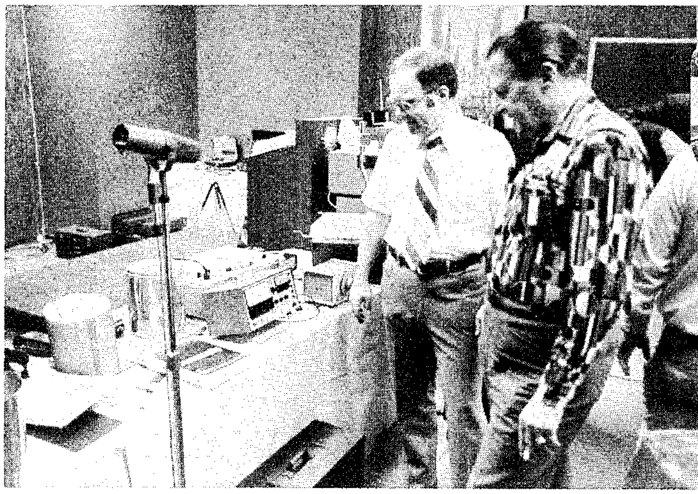
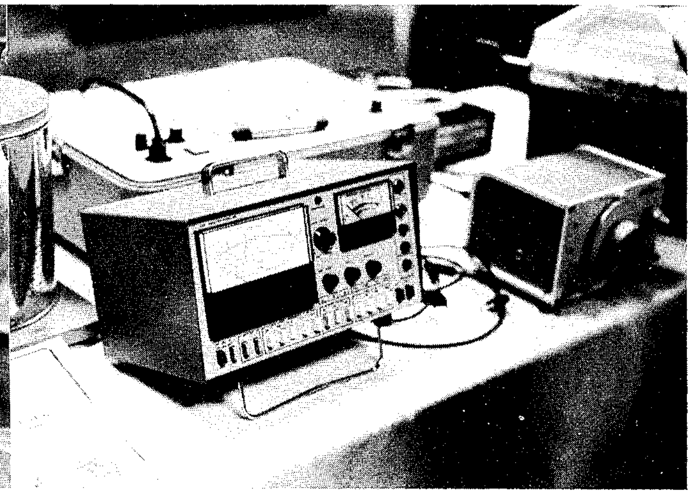
ED MODEL 400 CONSTANT-VOLTAGE SYSTEMS ANALYZER

With the demise of the Sennheiser "Scheinwiderstandsprüfer" (I'll bet the Craig Language Translator would choke on that one), Syn-Aud-Con has had one eye peeled for a suitable replacement impedance meter.

DAVE MOORE of Electro-Com in Seattle brought the new Electro-Dynamics Model 400 Constant-Voltage Systems Analyzer to the special Rauland class in Anaheim in December (we stopped counting the number of times Dave has attended a Syn-Aud-Con class when it went past 4).

Features we feel are particularly important are:

1. Wattage readings up to 1kw
2. Either 5 internal oscillator frequencies or *plug in* external oscillator
3. Battery or ac powered
4. A separate ground fault circuit is included for rapid checking of shorts from either speaker line to ground (from a dead short to over 100K ohms)
5. Low price of $539.

We tested the ED-400 in the Rauland-Anaheim class using both the internal oscillator and our GR 1309. That's DAVE MOORE (on the left) with LOUIS STOCKARD. The accuracy of the ED-400 is excellent and Dave's speed in finding and reading impedance peaks on our loudspeaker were practically instantaneous.

A well-built, very attractive unit, we are planning on trying it out in a series of our regular classes and should have more to report on it in the future. Our present judgment is that it is a superior choice over any we have suggested previously.

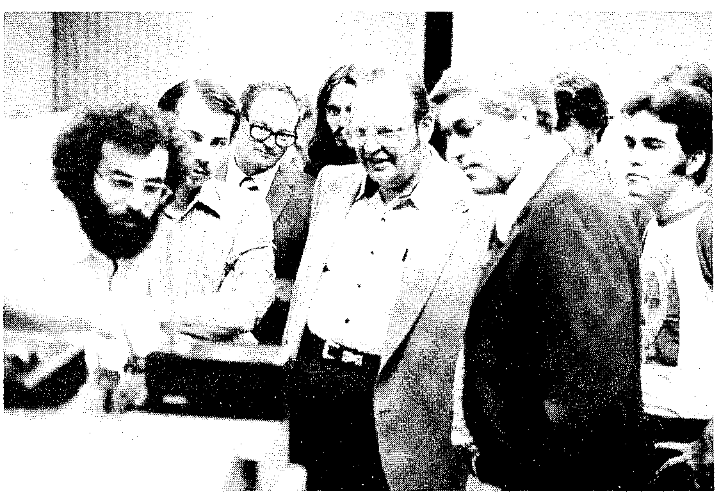
You can write Wayne A. Pommer, Electro-Dynamics Development Co., 475 Mt. Hood Dr. SW, Issaquah, WA 98027. (312/392-2493)

NEWSLETTER CORRECTION

On page 23 of Newsletter Vol 6 # 1 a typographical error appears in the equation in the article entitled, "Using the Available Power Concept".

It should have read: Available power in dBm = $10 \log\left(\frac{E_O^2}{.001Z_S}\right) - 6$ dB. We left "-6 dB" out of the equation.

CONNECTING THE IVIE IE-30 RTA TO THE NSL SCOPE



FARRELL BECKER is showing part of the class in Orlando how he wired the external outputs of the IE-30 RTA to the NLS dual trace oscilloscope.

The top trace is a standard time domain oscilloscope display.

The bottom trace is a frequency domain RTA display.

Mental FFT comparisons are inevitable with this hookup.

HAPPY BIRTHDAY DECIBEL

January 1979 is the fiftieth birthday of the decibel. It was born in W. H. Martin's article, "Decibel - the Name for the Transmission Unit" in the January 1929 Bell System Technical Journal, pages 1 & 2.

As this most helpful tool passes into its second half century of usefulness, we can only wish "more power to it".

AN EXERCISE IN ADDING DECIBELS

$$\text{dB-SPL}_D = \text{dB-PWL} + 10\log\left(\frac{Q}{4\pi r^2}\right) + 10.5$$

Where: dB-SPL_D is the *direct* sound level ref. to 20 μpa

dB-PWL is the acoustic power in watts expressed in decibels ref. to 10^{-12} watt

$$10\log\left(\frac{\text{acous. power in watts}}{10^{-12}\text{ watts}}\right)$$

Q is the directivity factor (dimensionless)

r is the distance from the sound source in feet

10.5 is a constant associated with the conversion of metric dimensions to English units

$$\text{dB-SPL}_R = \text{dB-PWL} + 10\log\left(\frac{4}{S\bar{a}}\right) + 10.5$$

Where: dB-SPL_R is the reverberant sound level ref. to 20 μpa

$S\bar{a}$ is the total absorption in ft^2 or sabins

The combination of these two equations results in the familiar Hopkins-Stryker equation.

Since dB-PWL is common to both equations because the *same* power source drives both the direct and reverberant sound fields, it need appear but once in the combined equation. The constant 10.5 is also common to both equations because of a single metric conversion, thus it also need only appear once in the combined equation.

Therefore, using the fundamental decibel combining technique, we can write:

$$\text{dB-SPL}_T = 10\log\left(10^{\frac{\text{dB-PWL}}{10}} + \frac{Q}{4\pi r^2} + \frac{4}{S\bar{a}} + 10^{\left(\frac{10.5}{10}\right)}\right) = \text{dB-PWL} + 10\log\left(\frac{4}{4\pi r^2} + \frac{4}{S\bar{a}}\right) + 10.5$$

Further examination of this technique reveals:

$$\text{dB-SPL}_T = 10\log\left(10^{\left(\frac{\text{dB-SPL}_D}{10}\right)} + 10^{\left(\frac{\text{dB-SPL}_R}{10}\right)}\right)$$

and

$$\text{dB-SPL}_R = 10\log\left(10^{\left(\frac{\text{dB-SPL}_T}{10}\right)} - 10^{\left(\frac{\text{dB-SPL}_R}{10}\right)}\right)$$

L_D 63.1200 ***
L_R 74.7100 ***
L_N 55.0000 ***
RT_{60} 2.5000 ***

%Al_{cons} 0.1945 ***

L_N 50.0000 ***

%Al_{cons} 0.1819 ***

L_D 69.0000 ***

%Al_{cons} 0.1353 ***

<u>A Useful Example</u>

In working with the newest Peutz computer program (See Tech Topic Vol 5 # 12) it is possible to directly calculate %Al_{cons} from the following parameters:

- dB-SPL_D
- dB-SPL_R
- dB-SPL_{amb} (ambient noise level)
- RT_{60}

Now, dB-SPL_T, dB-SPL_{amb} and RT_{60} are directly measurable. dB-SPL_D can be measured at some point (at least 10 dB above the reverberant sound level) near the sound source and extrapolated via the inverse square law to the desired measurement point. If, for example, we were to measure at 125 feet a dB-SPL_T = 75 dB in an auditorium with an RT_{60} = 2.5 secs and a dB-SPL_{amb} = 55 dB and we then found that at 8' from the sound source we had 87 dB-SPL_D, we could then find the dB-SPL_D at 125' by

$$87 + 20\log\left(\frac{8'}{125'}\right) = 63.12\text{ dB dB-SPL}_D$$

and we could then further find the dB-SPL_R by

$$10\log\left(10^{\left(\frac{75}{10}\right)} - 10^{\left(\frac{63.12}{10}\right)}\right) = 74.71\text{ dB-SPL}_R$$

Using these values in the Peutz computer program we obtain 19.45% Al_{cons}

Changing the dB-SPL_{amb} to 50 dB, we then obtain 18.19 %Al_{cons}; thereby find out that our direct sound level needs to be raised as well. Trying 69 dB-SPL_D, we get 13.53 %Al_{cons}. This now tells us that we need to increase our Q by a factor of

$$10^{\left(\frac{69-63.12}{10}\right)} = 3.87$$

in order to improve our direct level from 63.12 dB to 69 dB.

DAVID CLARK HEARING PROTECTORS MODEL 27-L

That CHIPS DAVIS is willing to take chances in the quest for progress is self-evident from his LEDE control room.

The photograph indicates that Chips has a physical prowess as well as good nerves. He is firing a 458 magnum and that's real recoil, not posed.

When men like Chips use these high output devices (174.5 dB-SPL) it is critical that they protect their highly sensitive and trained hearing.

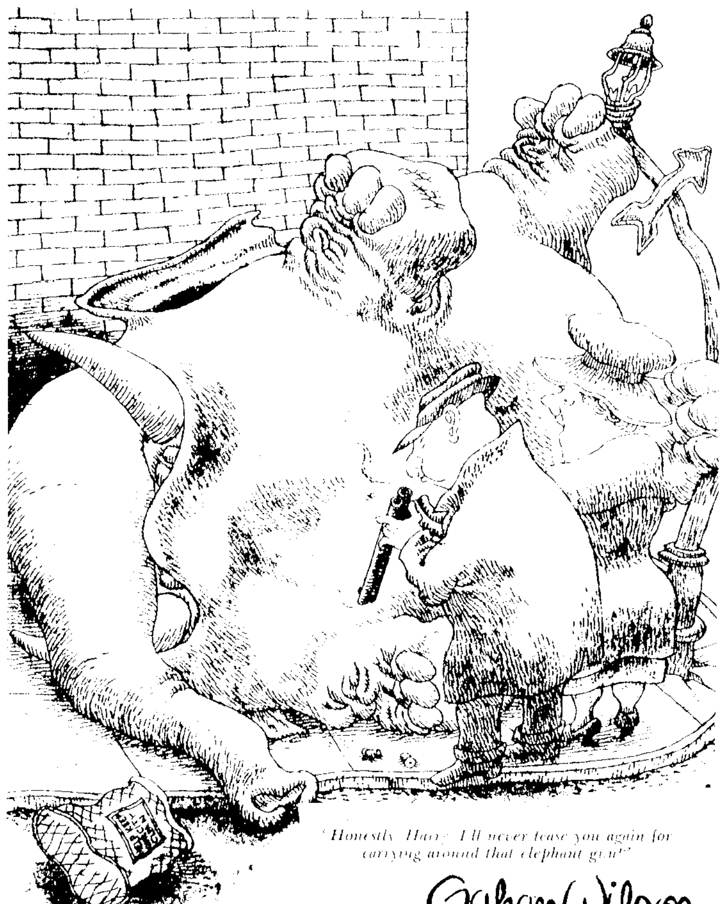
The hearing protectors Chips is wearing - the new David Clark model 27-L - actually reduce these levels to a perfectly safe and comfortable figure. We both wish there was a recoil pad as effective.

The drawing below illustrates how each of us is prepared to protect ourselves from unexpected urban hazards.

GLENN BARKER of WED Enterprises heard one too many of my elephant stories during the 1978 Los Angeles class and felt I deserved it.

On occasion people have wondered why Bill Raventos of of Ivie Electronics, CHIPS DAVIS, KEN WAHRENBROCK and myself go out of our way to shoot rifles that "kill at both ends". (Capital letters just means that they have attended a Syn-Aud-Con class)

Bill Raventos is using a 460 Weatherby magnum (the most powerful rifle commercially available in the world today) and the rest of us are using the 458 Win. magnum (the second most powerful). An army Garand kicks the shooter with 19 ft lbs of recoil. The 458 mag. manages to raise this figure to 68 ft lbs. and Bill is experiencing 105 ft lbs. (Also ask Bill sometime to tell you about the rattlesnake bite he received while hiking alone two days into the mountains.)



"Honestly Harry, I'll never tease you again for carrying around that elephant gun!"

Gahan Wilson

How Caruso Shattered Wineglasses

Since this year marks the 100th anniversary of the birth of Enrico Caruso, the news media have revived many of the anecdotes in connection with this colorful and gifted artist. One of these stories relates to Caruso shattering wineglasses by means of his voice. The purpose of this piece is to provide some pertinent information on this and allied subjects relating to Caruso's powerful voice.

While attending the University of Iowa in the middle 1920's, Professor Baker corroborated the story of Caruso shattering a wineglass by means of his voice. A wineglass consists of a foot connected by a stem to a variously shaped cup holding four to six ounces. He said that Caruso selected cheap glasses with high internal stresses and strains. Such glasses are inherently fragile, particularly when the integrity of the surface has been violated. The "tempered" glasses of today are much tougher. He scratched the outside surface of the glass with his diamond ring to facilitate the shattering. He determined the resonant frequency of maximum amplitude by touching the bottom of the bowl very lightly with his fingers while sounding a continuous note through a frequency range. In this connection, the exciting frequency was not necessarily the fundamental. When the resonant frequency had been determined, he sang this note as loud as possible until the wineglass shattered. (Professor Baker is Professor of Mathematics, at Iowa.)

A spectrum analysis of Caruso's records, corrected for the frequency response of the recording characteristic, indicates that the maximum output occurs in the frequency range of the maximum amplitude response of wineglasses. This appears to be a requisite for anyone to be able to accomplish this feat.

In 1927, some of my classmates and I, in recalling Professor Baker's story, became interested in shattering wineglasses by means of the voice. In spite of Prohibition, we obtained some cheap glasses from the "5 and 10 cent" store, and scratched these glasses with a glass cutter. Two out of a group of twenty who experimented were able to shatter some of the glasses.

In 1933, the RCA Laboratories were moved from New York City to Camden, New Jersey. Up to this time and a few years following, the main recording facilities of RCA Victor Records were located in Camden. Since the mechanical recording equipment was still available, I became interested in determining the sound spectrum and sound level of Caruso's voice in the recording of records. From the maximum amplitude of the recording, and the distance of Caruso from the mouth of the recording horn, the maximum sound level at two inches from Caruso's lips turned out to be 140 dB. The engineers who had recorded Caruso said that he could sing much louder, since he was under no strain during recording.

Aside from the artistic aspects of Caruso's voice, there are some features which conspired to make his voice particularly suitable for, and complementary to, the limitations of mechanical sound recording and mechanical sound reproducing, that is, recording and reproducing sound without electronic assist.

Being a tenor, his low frequency limit was about 150 Hz. The first few partials in his voice were very powerful, meaning that the high frequency range was not so important. The major power of his voice was confined to a frequency range of 150 to 1500 Hz. In this frequency range, the fidelity of mechanical sound recording and reproducing systems is quite good. In addition, Caruso could produce full amplitude on the record without any particular effort, making the sound output in mechanical sound reproduction of a very high level, and contributing markedly to the artistic impact in the reproduction.

In summary: the high sound power output, the frequency range and spectrum distribution of Caruso's voice cooperated to produce the maximum possible fidelity with the mechanical sound recording and reproducing system and thereby provided the greatest artistic impact with a system of inherent technical limitations. The high sound power output, a particularly suitable frequency power characteristic and the fine pitch control of Caruso's voice enabled him to shatter wineglasses by this means.

HARRY F. OLSON

IDEAL POLAR PATTERN

Recently, as the result of a consulting assignment, we recommended that the sound engineers responsible for a church sound system test several of the new high Q horns available. The church was long, narrow, high ceilinged, and very reverberant.

In making the tests an Altec Manta Ray, a JBL unit, and the EV high Q horn were tried and carefully listened to. All three units approached a Q of 50 in the 2,000 Hz octave band.

The results of these tests were different than we expected, but upon reflection revealed something obvious. It should be noted that these tests were conducted in a reverberant auditorium geometrically suited to all three horns.

What the engineers found was that the Altec and EV horns literally "cut off" too sharply and left them with inadequate levels at the fringes of their coverage patterns. The JBL having a more gradual cut off of its coverage pattern allowed sufficient energy to strike the side walls and assist articulation but not enough to allow a significant increase in the reverberant sound field level.

This has led to our contemplation of rates of change in polar responses and we could see that the same rate of change from on-axis to C_L beyond C_L to $2C_L$ should be used. Beyond $2C_L$ the rate can become whatever the horn designer is capable of making it.

We would be interested in hearing from graduates as to what their experiences have been along this line of reasoning.

ECD CORP

Of interest to Syn-Aud-Con graduates is: ECD Corporation, 232 Broadway, Cambridge, MA 02139.

They make a series of unique products among which are digital capacitance checkers (direct LCD readout and automatic ranging), temperature measurers and much more. Our experience with their Model 100 Capacitance checker leaves us with no reluctance to recommend their products. You can write for their current catalog.


RAULAND/ORLANDO CLASS
Bob de Freese
Eugene K Zorn

TED L. LeTOURNEAU's father was the inventor of the large earth moving machinery that have so changed our lives. The machinery used powerful electric motors in the hubs with electricity for them being generated by a large diesel engine and generator. Thus Ted has experience in "moving mountains". Moving the mountains of misinformation and ignorance standing in the way of better listening conditions in churches takes a "prime mover" such as Ted gave us when he attended an Update class in Orlando. (Ted has an acoustical consulting firm in Longview, TX)

HEARING!!!

HOW DID JESUS TALK TO SO MANY AT ONCE...
A NEW LOOK AT THE PARABLE OF THE SOWER

It is recorded in Matthew, Chapter 13, that Jesus taught a multitude of people. As on other occasions He went into a ship and the people stood on the shore. Only in the recent years of enlightened scientific understanding can we see all the advantages this gave him, to let hundreds or thousands of people hear our voice without electronic means.

The higher frequencies of sound radiate from a preacher's mouth in a conical pattern, like the shape of a megaphone. These are the sounds that carry the intelligence in speech, the consonant sounds. As Jesus talked, the sounds that radiated downward were reflected upward by the water. These along with sounds that originally radiated upward were bent back down again by the air being cooler near the water's surface and warmer above. This made a potent concentration of all the power of His voice into a thin horizontal spread that just covered all the people on the shore. Also there were no jet planes, trains, trucks, air conditioning, etc. So only a low level of sound was needed at the listener's ears, for them to hear clearly.

In a modern auditorium, these conditions do not exist. Jesus may have had a bulkhead behind him, (which would only help reflect the low vowel sounds) but, in these times we make a box by wrapping three more walls around the scene and putting a roof over it. This causes all kinds of problems with reverberation, echos, and noise.

We build a *long* box and the sound diminishes to the distant listeners. We build a *wide* box and the preacher cannot direct the intelligence carrying consonant sounds to everyone at once. So some seeds fall by the wayside. (V-4).

We build a *large* box to seat more people and even put a powerful amplifier system in it. The sound may be loud enough to be heard by everyone but only vowel sounds come through while the effectiveness of consonant sounds are destroyed by reverberation. So some seeds fall on stony places. The sound is all there but the heat (V-5,6), of reverberation withers away the intelligence.

We often put in noisy air conditioning or build on a heavily traveled street. We might even poorly design the building and have echoes that flutter back and forth. We also have people in our congregations sometimes, who thoughtlessly create noise and disturbances. So some seeds fall among thorns and thistles which spring up and choke the sound with background noise too loud to be overcome (V-7).

Today only occasionally, we do plan and build our church auditorium with thought in mind from the very beginning, that it should be a place to *hear*. Our church should be a place where everyone can hear clearly, every word spoken by the preacher. It should be a place where music can "ring to the house top" because people like to sing there. So other seeds fall on good ground and bring forth fruit (V-8).

It is too bad (my greatest concern) that many church organizations spend hundreds of thousands of dollars to build beautiful churches in order to attract unsaved people to come in and hear the gospel, but when they do come in, they can't hear! Why not let them hear? (V-9). And, that's not the half of it. The situation is much worse than you may think.

But first, let me explain a little known principle. God gave each of us a wonderful computer which we carry around in the top half of our heads. We, who have accepted Jesus Christ as our own personal savior, have been attending church, often, several times a week. We have been listening to the gospel and Bible stories, doctrine and illustrations for some time. All this time we have been programming our computer and storing information in its memory banks. As a result we now have the ability, when we can't hear perfectly, to have our computer fill in the blanks for us. If a syllable or two, or even whole words now and then, are not understood when they are spoken, our trusty computer goes to work (but, only if we want to hear and are concentrating). It then searches our memory banks for a similar context of material, or even deduces from words following what we should have heard at that point, and fills in the blank just as though we had actually heard that syllable or word correctly the first time. We may not even have been aware that this has happened, except that after long periods of listening in this condition, we feel fatigued because of the mental effort required.

Now, here is the problem: That unsaved person who does come in (for whatever reason) and listens to the music; and hears the spoken word; has not programmed his computer as we have. In the first place, he probably is not accustomed to hearing the language used in church; and "King James" pronunciations; or illustrations taken from unfamiliar Bible stories or doctrine, especially when there is no explanation given. In the second place, he has no idea how to expect what will be said next. And, last but not least, because he "shall hear and shall not understand...shall see and shall not perceive," (V-14). He certainly won't put forth the mental effort of concentration needed to *try* to fill in the hearing blanks, when he is so poorly equipped to do so.

Therefore: What *we* may allow as an acceptably low level of lost consonants in speech, may not be at all tolerated by the unsaved person. Because to him it sounds like "just so much gibberish."

How then, can we expect the Holy Spirit to work in the mind (or heart) of the listener, when the Word of God never reaches the heart? For, faith comes by *hearing*, and hearing by the *Word of God.*

Is it good stewardship to spend God's money for comfortable pews without spending as much for the clear, comfortable listening that modern technology has made available in the 1970's? Will the lost be saved with only a comfortable place to sit and pleasant company???

"To see" implies to understand. "To hear" implies to obey. Do we obey verse nine--"Whoever has ears to hear *let* him hear." (Make it *possible* for him to hear) so that "hearing, they should understand with their heart, and should be converted, and should be healed." (V-15).

UZZLE UTTERANCES

When the roof of the Hartford Civic Center collapsed, Newsweek described it as "computer designed". The editor of Machine Design retorted, "a statement such as that could only appear in an article written by typewriter."

***

Shout "Movie!" in a Crowded Firehouse Department: lay apprehensive hands on the May number of Radio-Electronics (49:5) and read Gorin, "Build Graphic Equalizer for Your Stereo System", pp 37-40. Do you and Carolyn think you have seen the concept of combining equalizers, with ripple-free combined response, spat and shat upon? See fig 3 in the main article, and especially Len Feldman's virture-of-necessity column on p 39 (has this man no integrity?). Seriously, for preposterous ignorance this article is something special, and you should definitely read it. It is true of acoustics what Kierkegaard said of philosophy: at every step it sloughs off a skin into which creep its worthless hangers-on.

COMMUNICATIONS FROM MEL SPRINKLE

When we see a letter from MEL SPRINKLE in the "in box" two conflicting emotions cross our thought:

Emotion # 1: Ye Gads! I've made another mistake in the Newsletter or Tech Topics.

Emotion # 2: Thank you, Mel, we'll get it right yet.

Mel often signs these missives, "O. Howitt Hertz"

Mel's expertise is circuit theory. He currently is teaching a course in advanced ac theory at Capitol Tech, an electronics school in the DC area. Students with access to his classes are fortunate indeed.

The thrust of his latest "editing" has to do with my admittedly careless use of the symbol Z when I mean the symbol R.

More importantly, he included some criticism of the question on the dB that appeared in the Newsletter, Vol 6, # 1, page 5. I'll let Mel's comments speak for themselves:

> On page 5 of Newsletter Volume 6, # 1, there is this problem for cogitation: "We are given a 'black box' and we find that if 2 volts appear across the input, a voltage of 1 volt appears across some unstated value of load connected across the other of two ports (presumably the output.) The question is, then, does the 'black box' act as an amplification or loss device?"
>
> The "official" answer in the issue is that it's like a woman's answer: "it depends". This states that "the input and output impedances must be known". You state that if the "input impedance" is 100 ohms and the "output impedance" is 10 ohms, then the device has 4 dB of "gain". This appears to be based upon the notion that the input resistance (not impedance) absorbs 40 milliwatts of power which is +16.02 dBm which the load resistance (again not impedance) receives 100 milliwatts of power which according to my trusty HP-9100B is +20 dBm. Thus the difference is 3.98 dB which is close enuf to 4 dB for Government work!
>
> Now Don, there are several clinkers in this reasoning which Olde Grandad would like to invite your attention to. These are to wit:
>
> a. The term "impedance", while not defined on page 276 of the Canon means the *total* electrical opposition to the flow of current - which imples that there are or can be either reactances, or resistance or both. Now, as you also well know (I am sure!) reactances, either inductive or capacitive, *cannot* dissipate power; only resistance being capable of power absorption and conversion to heat. Thus if the "black box" in its input had any reactance, either in shunt or in series, then the "input impedance" can and will be different from the resistance component. If the reactance is in shunt with an input resistance, then it can be converted into an equivalent series impedance with resistive and reactive components. In such a circuit the resistive component is different from the shunt case (where R and X are in parallel) and also the input "impedance" is frequency dependent.
>
> b. You might also refer to page 276 of the Yellow Canon where the term "Impedance, output" is defined. I invite your attention to the footnote (always read the fine print in any document) which states that the term is sometimes *incorrectly* used to designate the load impedance. Thus your use of this term on the "official answer" refers to the Thevenin source impedance of the output circuitry or output port of the port of the "black box". Thus it cannot be that the output is 100 milliwatts because the output impedance of 10 ohms refers to the Thevevenin source impedance and we cannot calculate the power dissipated therein until we know: (a) the output port current, which also depends upon the "impedance, load" (also defined on page 276 of the Canon) and (b) what is the resistive component of the "Impedance, Output".
>
> c. Aside from the incorrect terminology, the "gain" represented by the ratio of input power absorbed to the power delivered into an "impedance, Load" is called "power gain" and is defined on page 278 of the Canon. Unfortunately, the "power gain" of an amplifier is of little interest and/or usefulness.
>
> I am attaching several pages reproduced from American National Standards Institute (ANSI) Standard C16.29-1957 which is the same as IRE-IEEE Standard 56-IRE-3.S1. This may be purchased from ANSI in New York, or is also in Proceedings of IRE, May 1956. I am encircling several passages which are of great interest and importance. Your attention is especially invited to the section on Transducer Gain in which the *available power* from a Thevenin Generator is so important. Also, the G_m(EIA) microphone rating as well as the "dbm" rating are all *available power*.
>
> Thus in order to properly and accurately define whether a "black box" is a gainer or loser you must state the Thevenin generator constants (open circuit voltage and source impedance) together with the "Impedance, Load" and whether the load has reactance. Also the output voltage across the load must be given.

Communications from Mel Sprinkle, continued

1. Gain, Amplification, Loss, Attenuation

In audio-frequency usage "gain" and "amplification" and the inverse terms "loss" and "attenuation" are general terms pertaining to the transmission characteristics of audio systems and components, and are commonly used to express significant power, voltage, or current ratios. In order to avoid ambiguity, it is recommended that the terms gain and loss be used only to express *power* ratios, while the terms amplification and attenuation should be limited to the expression of *voltage* or *current* ratios. Gain and loss are generally stated in decibels. Because of long established usage in the audio field, amplification and attenuation ratios are expressed in terms of decibels in this standard, although the more recently introduced term "decilog" is more appropriate. If the use of the terms attenuation and amplification is restricted to the expression of the results of measurements of *voltage* or *current* ratios, much of the confusion that has existed in the past usage will be avoided.

1.1 Gain (Transmission Gain)

Gain (transmission gain) is a general term used to denote an increase in power in transmission from one point to another. Gain is usually expressed in decibels, and the term is widely used in audio practice to denote "transducer gain."

The measurement of several types of gain is outlined under specific headings in the material which follows. In every case gain is a ratio of powers. The type of gain to be measured will depend on the desired application. In each instance it is recommended that the stated result be identified by the full terminology (*e.g.*, Transducer Gain rather than Gain) unless, by context, or otherwise, there appears to be no possible chance of ambiguous interpretation.

1.1.3 Power Gain

Power gain is the ratio of the power that a transducer delivers to a specified load, under specified operating conditions, to the power absorbed by its input circuit. This gain is usually expressed in decibels.

The power gain of an audio system or component is ordinarily of rather limited interest. A statement of the power gain of a particular audio system or component usually does not give the most useful information regarding the transmission gain which will result if this system or component is inserted in or bridged across a second audio transmission system. For example: The input impedance of a microphone preamplifier is often very high compared to its specified source impedance. The power gain of the preamplifier may approach infinity, but this is not significant since the more useful gain of this amplifier is its transducer gain. (See section 1.1.4.)

1.1.3.1 Power Gain Relations

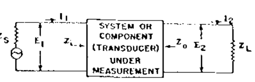


Fig. 3—Elementary circuit for power gain

$Z_i = R_i \pm jX_i$ = Input impedance of system or component.
$Z_L = R_L \pm jX_L$ = Load impedance.
$Z_o = R_o \pm jX_o$ = Output impedance of system or component.
$Z_S = R_S \pm jX_S$ = Source impedance.

$$\text{Power Gain in db} = 10 \log \left[\frac{(I_2)^2 R_L}{(I_1)^2 R_i}\right]$$

Since

$$I_1 = \frac{E_1}{Z_i} \quad \text{and} \quad I_2 = \frac{E_2}{Z_L},$$

$$\text{Power Gain in db} = 10 \log \left[\left(\frac{E_2}{E_1}\right)^2 \left(\frac{Z_i}{Z_L}\right)^2 \frac{R_L}{R_i}\right]$$

If Z_i and Z_L are pure resistances R_i and R_L respectively, the above equations can be simplified as follows:

$$\text{Power Gain in db} = 10 \log \left[\frac{(E_2)^2 R_i}{(E_1)^2 R_L}\right]$$

This equation is rigorously correct even if reactance is present in the input impedance Z_i and the load impedance Z_L provided R_i and R_L represent the equivalent parallel resistance of Z_i and Z_L respectively. Equivalent parallel resistance R_{par} can be computed from the series resistance R_{ser} and series reactance X_{ser} by the expression:

$$R_{par} = \frac{(R_{ser})^2 + (X_{ser})^2}{R_{ser}}$$

1.1.3.2 Power Gain Measuring Circuits

The circuit of Fig. 3 can be considered as a measuring circuit for power gain by addition of the meters necessary to determine the magnitudes of $\mathbf{E}_1$ and $\mathbf{E}_2$, or $\mathbf{I}_1$ and $\mathbf{I}_2$. If the impedances of the meters are such as to affect the quantities being measured, proper allowance must be made. For information concerning the required characteristics of meters, reference should be made to section 1.5.

1.1.4 Transducer Gain

Transducer gain is the ratio of the power that a transducer delivers to a specified load under specified operating conditions to the available power of a specified source, and is usually expressed in decibels. Stated in another manner, transducer gain is the ratio of the power that a given transducer delivers to a specified load from a specified source to the power which would be delivered if the actual transducer were replaced by an ideal transducer.[2]

Transducer gain (or loss) is the most frequently employed measure of the transmission characteristics of audio systems or components. The transducer-gain concept is so widely used and so firmly established that it is often referred to loosely as simply "gain" (or "loss"). In order to avoid possible ambiguities, however, it is strongly recommended that the full terminology (*i.e.*, "transducer gain" rather than "gain") be used to express the results of measurements of transducer gain.

1.1.4.1 Transducer Gain Relations

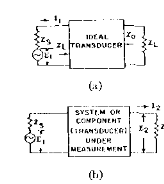


Fig. 4—Elementary circuit for transducer gain. (a) Circuit for determining reference power with ideal transducer. (b) Circuit for determining load power with system or component substituted for ideal transducer.

$Z_S = R_S \pm jX_S$ = Source Impedance
$Z_i = Z_S^* = R_S \mp jX_S$
$Z_L = R_L \pm jX_L$ = Load Impedance
$Z_o = Z_L^* = R_L \mp jX_L$

Since by definition an ideal transducer dissipates no energy, the reference power in Z_I of Fig. 4(a) above must equal the power in Z_i. Thus the power in $Z_I = (I_1)^2 R_i = (I_1)^2 R_S$.

Note that the reference power is the maximum available power of the source.

Then:

$$\text{Transducer Gain in db} = 10 \log \left[\frac{(I_2)^2 R_L}{(I_1)^2 R_S}\right]$$

or, since

$$I_1 = \frac{E_1}{2R_S} \quad \text{and} \quad I_2 = \frac{E_2}{Z_L}$$

$$\text{Transducer Gain in db} = 10 \log \left[\frac{4(E_2)^2 R_L R_S}{(E_1)^2 (Z_L)^2}\right]$$

[2] **Ideal Transducer** (for connecting a specified source to a specified load). A hypothetical passive transducer which transfers the maximum possible power from the source to the load.
Note: In linear transducers having only one input and one output, and for which the impedance concept applies, this is equivalent to a transducer which (a) dissipates no energy and (b) when connected to the specified source and load presents to each its conjugate impedance. (*54 IRE 3 S1 Standards on Audio Techniques, Definitions of Terms, 1954;* Proc. IRE, vol. 42, pp. 1109–1112; July 1954.)

If Z_L is a pure resistance R_L, the above equation can be simplified as follows:

$$\text{Transducer Gain in db} = 10 \log \left[\frac{4(E_2)^2 R_S}{(E_1)^2 R_L}\right]$$

This equation is correct even if reactance is present in the load impedance Z_L, provided that R_L represents the equivalent parallel resistance of Z_L.

1.1.4.2 Transducer Gain Measuring Circuits

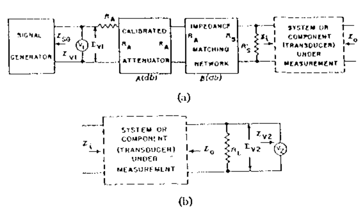


Fig. 5—Suggested circuits for gain measurement. (a) Source circuit arrangement. (b) Load circuit arrangement. (c) Alternate load circuit arrangement.

A—Transducer loss in db of the source-circuit calibrated attenuator.
B—Transducer loss in db of the source-circuit impedance-matching network when operated from a source impedance R_A into a load impedance R_S.
C—Transducer loss in db of the load-circuit impedance-matching network when operated from a source impedance R_L into a load impedance R_D.
D—Transducer loss in db of the load-circuit calibrated attenuator.
$\mathbf{E}_{V1}$—Voltage output of signal generator, as indicated by voltmeter V_1.
$\mathbf{E}_{V2}$—Voltage developed across R_L, as indicated by voltmeter V_2.
R_A—Iterative impedance of source-circuit calibrated attenuator; and input impedance of source-circuit impedance-matching network when the network is terminated in R_S.
R_D—Iterative impedance of the load-circuit calibrated attenuator.
R_D'—Resistance equal in value to R_D.
R_L—Load resistance of system or component under measurement.
R_S—Output impedance of source-circuit impedance-matching network when output terminals of audio signal generator are short-circuited.
R_S'—Shunting resistance across output of source-circuit impedance-matching network.
V_1—Source voltmeter.
V_2—Load voltmeter.
Z_i—Input impedance of system or component under measurement. May contain both resistance (R_i) and reactance (X_i).
Z_o—Output impedance of system or component under measurement. May contain both resistance (R_o) and reactance (X_o).
Z_{SG}—Output impedance of signal generator.
Z_{V1}—Impedance of voltmeter V_1.
Z_{V2}—Impedance of voltmeter V_2.

The circuit of Fig. 4(b) can be considered as a measuring circuit by the addition of the voltmeters necessary to determine the magnitudes of $\mathbf{E}_1$ and $\mathbf{E}_2$. If the impedances of the voltmeters are not sufficiently high so as to have negligible loading effect upon the circuit under measurement, proper allowance must be made.

In some instances it may prove impractical from the standpoint of operational accuracy or convenience to make measurements directly as described above. Therefore, suggested measuring circuits representative of good engineering practice and adaptable to a variety of gain measurements are shown in Fig. 5(a) and 5(b).

These circuits are particularly applicable when the reference source and load impedances of the audio system or component under measurement are resistive in nature, which condition obtains for the great majority of measurements required in audio engineering practice. However, these same circuits may also be used in cases where the source and load impedances of the system or component under measurement are reactive, providing the source-circuit and load circuit impedance-matching networks are suitably modified and calibrated at the measuring frequency.

The source circuits shown in Fig. 5(a) are assumed to include any requisite isolating transformers, as well as appropriate provision for the grounding of the audio system or component under measurement. The source-circuit impedance-matching network of Fig. 5(a) is required when the source-circuit calibrated attenuator does not provide the desired source impedance for the equipment under test.

The load circuits, Fig. 5(b) or 5(c), likewise are assumed to include any isolating transformers and grounding provisions required by the audio system or component under measurement. If the impedance Z_{V2} of Fig. 5(b) is not so high that its shunting effect on R_L can be neglected, then this impedance must be included as part of R_L.

The ability of the measuring circuit to accommodate a wide range of output levels and impedances may be increased by the addition of an impedance-matching pad and a second calibrated attenuator in the load circuit of the system or component under measurement. This is shown in the alternate load circuit of Fig. 5(c). In typical measuring practice, several elements shown in the circuits of Fig. 5(a) through 5(c) may frequently be combined in one physical instrument.

For the measurement of transducer gain the following special conditions apply to the circuits of Fig. 5(a) and 5(b):

$R_S' = \infty$.

R_S = Specified source resistance of system or component under measurement

R_L = Specified load resistance of system or component under measurement

MANFRED SCHROEDER

Several years ago we reviewed *Auditorium Acoustics* in a Syn-Aud-con Newsletter (Edited by Robin MacKinzie, published by Applied Science Publishers Ltd., London, 1975)

At that time we completely overlooked the fact that Manfred R. Schroeder's work on two channel surround sound and of equal importance his work on diffusion was included in the book. We have more on his diffusion technique elsewhere in this Newsletter.

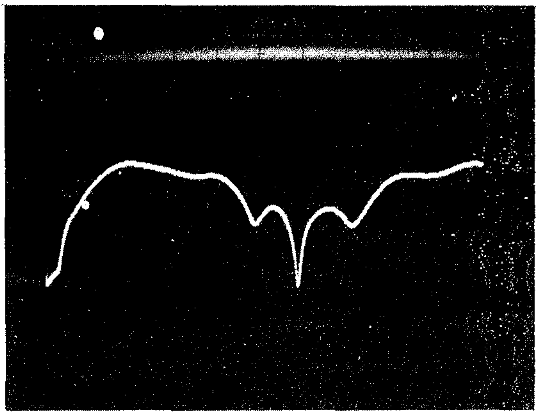
We had the privilege of hearing Schroeder discuss his diffusion technique at the ASA meeting in Honolulu in Dec. The sound system being used was absolutely atrocious, with a constant ring in the system. The nearest loudspeaker to the talker was about 50 feet. The loudspeakers, (3) 9844s painted pink, were not aimed at the listeners but at each other high in a ceiling with a soffit around the edges. The microphone was on the end of a long gooseneck extension.

I have always heard it said that in a desperate fight the man with a smile is one to look out for - well, with that same kind of graveyard smile, Schroeder grasped the microphone in a "death grip" and bent it and the gooseneck down to the floor with a rending and screeching of the protesting metal. It was at this point that technician ran for a lavalier. While the quality remained terrible, the acoustic gain was then enough to at least understand what was being said.

We came away with the feeling that Mr. Schroeder is not a man to be trifled with.

DIFFERENCE BETWEEN PHASING AND POLARITY

In current classes we are using a pair of Visonik "Little David"miniature loudspeakers to demonstrate the difference between "phasing" and "polarity".



The photograph taken with TDS measurements shows what a displacement of approximately 1" between these two identical loudspeakers causes in the response curve. (Sorry that we didn't take a picture of the Little David set up but the two speakers are stacked on top of each other with the top speaker moved back 1" out of line with the bottom speaker.)

A wavelength of one inch is a frequency of

$$f = \frac{V}{W} = \frac{1130}{1/12} = 13{,}560 \text{ Hz}$$

Therefore, the first deep notch (over 30 dB deep) occurs at

$$\frac{f}{2} = 6780 \text{ Hz}$$

Suppose the notch occurred at 5800 Hz. How far apart would the two sources be?

$$D \text{ in inches} = \frac{1130(12)}{2(5800)} = 1.17"$$

The general case equation being

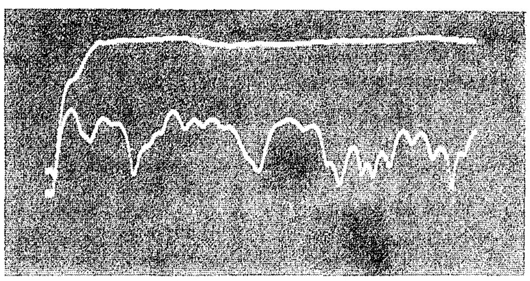
$$D \text{ in inches} = \frac{12V}{2(f_n)}$$

Where V is the velocity of sound in ft/sec

f_m is the frequency of the first notch (on the TDS this is usually $\frac{f}{2}$)

In arrays extreme care should be taken to minimize multiple drivers sharing *the same coverage area and frequency range*. The crossover network when properly designed reduces the alignment problem between two units covering *different frequency ranges* to a single notch.

POLARITY



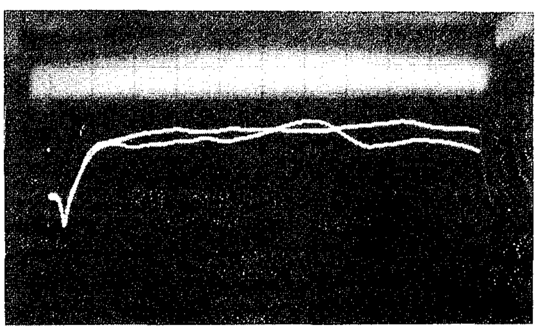
The difference between polarity and phase was discussed in the October 1978 Newsletter, Page 23.

Remembering that polarity is not frequency dependent, the picture shows some 40 dB of cancellation between two loudspeakers (bottom response) compared to their very uniform combined response (top trace).

Of interest to us is the narrowness of the area which exhibits good cancellation. Just an inch away from precisely inbetween the two units results in less than 6 dB of cancellation. This suggests that anytime it is desired to place another transducer in the "null" between two loudspeakers, great care should be taken to insure that the actual center has been found. Obviously a single sine wave tone is not sufficient. Either a TDS sweep or perhaps pink noise should be used

THE VISONIK "LITTLE DAVID" SPEAKERS

That you must continue to test even satisfactory products is borne out by the photograph of the Visonik "Little David" units.



0 - 20,000 Hz

The Visonik "Little David" units we originally purchased exhibit a "flat" response to over 20,000 Hz on the TDS. During one of the Fall classes we inadvertently damaged one of our two units when the power amplifier was connected to it while the amplifier's gain was adjusted for a larger unit.

We had a chance to compare our remaining "good unit" with a brand new "Little David" and found that the new units were noticeably less uniform in response (old unit is top curve and the new unit is bottom curve with peak around 6000 Hz passing through top curve and then back below top curve again).

Inspection of both units revealed that the old H.F. unit was a "hard dome" tweeter and the new unit had a "soft cone" tweeter. Why did they change? The older unit while smoother in response burnt out easier than the new unit. Small solace to those of us using them for tests rather thanlistening to rock concerts.Consequently our original damaged unit is back for repairs (they still have a supply of "hard dome" tweeters for repair purposes).

ACOUSTIC GAIN RELATIONSHIPS

ED LETHERT of Electronic Design in St. Paul has shared an interesting insight in acoustic gain relationships:

I found my interest in the acoustic gain equations stimulated during class and in trying some examples found something curious. I used Don's "favorite" room to try a comparison between the equation that used ΔDx and the one that uses the actual distances. Here are the results:

$V = 500{,}000\ ft^3$ $\quad D_1 = 40'$
$T = 2.5\ sec$ $\quad D_2 = 125'$
$S\bar{a} = 9{,}800\ ft^2$ $\quad EAD = 8'$
$Q = 9$ $\quad NOM = 2$
$D_c = 41.87\ ft$

Solve for D_s

Using the Hopkins-Stryker equation yielded the following:

$\Delta D40' = 30.68$ $\quad \Delta D8' = 19.36$
$\Delta D125' = 33.43$ $\quad 10 \log NOM = 3.01$

$\Delta D_s = \Delta D_1 + \Delta D(EAD) - \Delta D_2 - 10 \log NOM - 6$
$\Delta D_s = 30.68 + 19.36 - 33.43 - 3.01 - 6$
$\Delta D_s = 7.60$
$D_s = 2.03$ ft.

Now, trying the direct equation and observing the rule $Dx > D_c = D_c$

$$D_s = \frac{D_1(EAD)}{2D_2\sqrt{NOM}}$$

$$D_s = \frac{40x8}{2x41.87x\sqrt{2}}$$

$$D_s = 2.70$$

In trying to find the cause of inequality between the methods I stared at the Hopkins-Stryker equation and the curve it produces for considerable time. Finally the light came on. $D_x > D_c = D_c$ compensates for the reverberant field in the room but does not take into consideration the addition of the direct and reverberant sound field. This causes an error which is greatest at critical distance and is on the order of 3 dB. This error can affect the answer by as much as 1.41 times or more.

The big question was - "How does one compensate to arrive at the correct answer?"

The answer was quite simple and therefore caused me some irritation because of the length of time it took me to arrive at it. To take into consideration the addition of SPL_D and SPL_R and translate the sum into a distance requires an equation which will provide an equivalent distance (I need a term which does not use the work "equivalent"), and here it is:

$$D_{xe} = \sqrt{\frac{1}{\frac{1}{D_x^2} + \frac{1}{D_c^2}}} \qquad (1)$$

I think you will see why the reciprocals and squares are required. To convert the results of the following examples into the correct distance, the following is used

$$D_x = \sqrt{\frac{1}{\frac{1}{D_{xe}^2} - \frac{1}{D_c^2}}} \qquad (2)$$

Using equation (1) we calculate D_e from each value

$D_e(40') = 28.92'$
$D_e(8') = 7.87'$
$D_e(125') = 39.70'$

Now we apply these numbers to the following equation:

$$D_s = \frac{D_1 \cdot EAD}{2 \cdot D_2 \cdot \sqrt{NOM}}$$

$$D_e = \frac{28.92x7.86}{2x39.70x\sqrt{2}}$$

$$D_e = 2.02$$

When we use equation (2) to calculate D_s:

$$D_s = 2.03\ ft.$$

You may wonder why go to all this trouble. Well, if you look at the present SS design program, the Hopkins-Stryker equation and its inverse use 47 steps of program memory. Look at the following list of steps (only 14) which does the same job:

LBL 6	1/x
SF 2	F?2
LBL 1	CHS
x^2	+
1/x	1/x
RCL_{Dc}	$\sqrt{x}$
x^2	RTN

This subroutine converts D_x to D_{xe} via label 1 and D_{xe} via label 6. It also reduces calculation time tremendously. Besides the shorter operating time, it leaves many more steps available for additional program capability.


RAULAND/ANAHEIM CLASS
RICK BLUNT POURED

THE M_A & M_E D_C MODIFIER

We have on occasion treated M_a as if it were a modifier of Q. Such usage, however, *is incorrect*. Why? Because M_a can and does change the acoustic power going into the room. It is this absorption of *power* that changes the level of the reverberant sound field, hence the D_C. Changing Q can also change D_c but not the reverberant level, only the direct sound level is affected.

M_e on the other hand changes D_C by leaving the acoustic power output of a source the same but increasing the direct sound level (by shortening D_2) therefore acting like a Q modifier.

The importance of all this is that M_a should always be associated with $S\bar{a}$ in equations such as the Hopkins-Stryker and M_e should be associated with Q.

$$\Delta D_X = 10 \log \left(\frac{QM_e}{4\pi (D_X)^2} + \frac{4N}{S\bar{a}M_a} \right)$$

Note that N also is a modifier of acoustic power in the space and that free field conditions are affected by

$$\frac{QM_e}{4\pi (D_X)^2}$$

whereas indoors reverberant conditions are affected by

$$\frac{4N}{S\bar{a}M_a}$$

That is, adding N essentially reduces the effect of $S\bar{a}$ and M_a increases the effect of $S\bar{a}$.

Much benefit can be derived from reviewing exactly how variations of basic equations operate physically in the environment.

LANGUAGE TRANSLATOR

Those fortunate to have been heavily involved in computers and electronic calculators for more than a decade have grown immune to the blandishments of many of the new offerings. My present programmable calculator is into its third year (is HP slipping??)

The latest announcement for a pocket size programmable unit really has caught our attention. It is the Craig translator.

This unit gives you a 50 phrase plus 7000 word vocabulary in four languages at a time - choose from English, French, German, Japanese, Spanish, and Italian. Up to four of these language modules may be plugged in at one time.

Twenty-five of the phrases are complete such as "How much does this cost?" and 25 are preliminary such as "May I please have the ________?" where you enter the word you wish. All this appears on the specially designed bright fluorescent blue display.

Future plans include additional languages and more complex modules for the languages already covered.

I would like to see a technical German module, as well as a technical Japanese, a Russian acoustical and mathematical module. Great lovers could send for undressed French. I still remember the German mechanic telling us about the absoluteness of the red line on our Porsche tachometer:

> "Und wen you reach zee red lines, you run around in back mit a bushel basket and catch the flying parts."

This Craig Translator sells through JS&A for $199.95 (2.50 shipping) plus $24.95 per language capsule. The English capsule is provided with the Translator. There are many features we lacked space to describe here but if you've ever wanted to communicate in a language you didn't know, this unit will be of interest to you.

NEWSLETTER CORRECTION FROM SHURE BROTHERS

BOB SCHULEIN, Chief Development Engineer - Acoustics from Shure Brothers found he could improve our knowledge of Shure microphones. We had written in Newsletter Vol 6, # 1 that "To our knowledge this (SM81) is the first condenser microphone that Shure has ever offered." Shure's first condenser microphone was the Model 40D introduced in 1935.

Shure's first modern condenser microphone is the SM-82, which was introduced in 1974. Is our face red! This is our favorite microphone system for testing purposes (has a built in line amplifier and peak limiter and can drive a power amplifier directly.) "When all else fails, read the instructions." It is an omnidirectional electret capsule. Our appreciation to Bob for catching this error and for the correct historical chronology.

AC, DC, OR NO-C

A series of coincidences have brought to our thought the scientist and inventor, Nikola Tesla. This Serbian genius born in what is now part of Yugoslavia (then part of the Austro-Hungarian border province of Lika) in the year 1856 decided the shape of things to come in the electrical world with finality when in 1888 he invented the polyphase induction machine. With the advantage of an ac system established and the availability of an electric motor to use ac electric power, it took just a short time to convert the earlier dc system to the 60 Hertz system we know today.

TED UZZLE, the Boston poet-philosopher, in some recent research on "the incunabula of the talkie" found the following fascinating data on the early history of electricity:

> In 1887 Edison hired Nikola Tesla, largely on the basis of Tesla's European reputation. He was an ardent champion of alternating current, and Edison was considering it for his generating station on Pearl Street in New York City. Immediately the two began rubbing each other the wrong way, as only two equally eccentric, equally brilliant men can do. One day Tesla was ticking off potential advantages of ac generation when Edison said, "There's fifty thousand dollars in it for you if you can do it!"
>
> The Croatian disappeared for a few months, and then invited Edison to his workshop. When the prototype generator successfully passed every test, Tesla asked for the fifty thousand dollars. Edison replied, "Tesla, you don't understand our American humor."
>
> Within a week Tesla was working for George Westinghouse, and within a couple of years Westinghouse was making motors and generators clearly superior to anything Edison had. Much later, when long distance power transmission became important, the game would be over. But remember, at that point most electrified buildings had the generator in a shed out back.
>
> The game would not end before a few low blows would be exchanged. In the middle of Edison's "safety" campaign against ac, he set up a dummy corporation with a retired clerk from the Edison works as president. In May 1889 this company bought three ac generators from Westinghouse, giving phoney uses to which they would be put. Apparently Edison made a gift of them to the State of New York. They finally surfaced on 6 August 1890, when William Kemmler was executed for murder, with a Westinghouse generator. The execution, not the murder. The publicity coup was brilliant, if grisly.
>
> In 1912 the Nobel Prize in physics was offered jointly to Tesla and Edison. Tesla, although desperately in need of the money and the recognition, wrote back that he would not share a prize, or a platform, or anything else with Mr. Thomas Alva Edison. The committee finally gave the prize to Nils Dalen, who promptly dropped back into the obscurity from which the prize had raised him.

The September 1978 issue of Audio Magazine has Canby reminiscing about the old Hotel New Yorker (where Carolyn and I attended our first Hi Fi shows back in the very early 1950s). We have reproduced a portion of the article:

Historical Horrors

And speaking of that, our friend Bert Whyte last year in his history of hi-fi shows missed out on (i e failed to mention) one of the prime marvels of the first Audio Fairs in the Hotel New Yorker over on 8th Avenue and 34th Street in the Big Apple That august hotel, vaguely out of the art deco era (1930s?) was indeed furnished with 120 volts of electric potential in each and every room, since it wasn't nearly old enough to boast gas light But thanks to dear old Thomas A Edison and his latter day namesake utility, now nicknamed Con Ed, the New Yorker was provided with the very best available electricity in the mid-town area — as we used to put it, 120 volts of d c current And we picked *that* for a hi-fi show! Maybe nobody realized it until the last minute.

But the show must go on, and the Audio Fairs did indeed Somebody with a proper electrical background had apparently foreseen the future and the lucrative possibilities in servicing Conventions at the hotel, electronic included There was a modest local alternator, somewhere down in the basement, which fed a c to extra outlets (right among the d c outlets) in — well, in some of the rooms Our earliest hi-fi exhibitors, therefore, had to choose the right socket, and if they did they were rewarded with that splendid cacophony which we now find so familiar in hi-fi shows the world over It was a brand-new sound, then, and not nearly as well appreciated as now If the wrong socket was chosen, in haste or by unfortunate accident, then there was smoke and damage and silence except for that ominous crackling hiss.

That alternator, if I remember correctly, was the handiest closing-time signal you can imagine Right on the dot, six o'clock or whatever, the a.c. failed, whereupon the constant and excruciating blasts of nonintegrated fi from dozens of superior sources suddenly ceased penetrating every wall into four surrounding rooms and into corridors all over And we were left in an instant with the loveliest silent d.c illumination you can envision The d c kept going — it was from Con Ed Conversations, continued, in low, hushed tones, and I'd guess there were more hi-fi deals made in those late moments of silence than in all the day's noise

All that goes back to one of T Edison's greatest failures — in the midst of his greatest triumph, the integrated electric light system as pioneered right there in New York at the famed Pearl Street power station and distribution system That system was built on direct current, two-conductor Very soon after, Edison worked out the three-conductor system but again via d.c When this idea was applied to its ultimately right area, the a.c distribution system, we had modern power — but old Thomas would not go along He stuck to d c and there were horrid stories perpetrated about people being fried via high-voltage a.c in those newfangled arrangements. This was made even more gory by virtue of the new electric chair, which fried but did not quite kill one of its earliest occupants. Result was that the older Eastern cities, notably Boston and New York (and I would suppose Philadelphia) became entrenched in the d c system and stayed that way an astonishing number of years, right on through the 1930s Of all the vicious impediments to fi, that of d.c in the home was surely — at that time — the worst Very nearly insurmountable I should know; I was there

John J. O'Neill's *PRODIGAL GENUIS - THE LIFE OF NIKOLA TESLA* published by Ives Washburn, Inc. (GARY WALLESEN of Aatronics in Boise, ID recommended the book to us early 1976 but didn't really read it until Ted Uzzle's letter came) is a completely uncritical biography of a man of genius who to all evidence available went mad in a socially acceptable way in the later years of his life. His short circuiting the entire output of the Colorado Springs power company through an experimental giant Tesla coil (called a magnifying transformer by Tesla and compared to astromonical telescopes) with accompanying cannon-like reports heard 15 miles away, and finally *burning out* the generator at the power plant is a story worth the price of the book (in paper back)

How like today where Nuclear power is handled in much the same scientific manner by the media. The results this time won't be dc where you need ac; it'll be "blackout" because we are falling inexorably behind the mandatory growth rate and replacement required of our power grid.

BOOKS OF INTEREST

ORIGINS IN ACOUSTICS by Frederick V. Hunt with a foreward by Robert Edmund Apfel, published by the New Haven and London Yale University Press 1978. $14.00

This 196 page book has a subtitle that accurately describes its virtues, "The Science of Sound from Antiquity to the Age of Newton."

Few scientists, and Frederick Hunt was the epitome of the word, can see past their formulas and dogma, but Hunt soars past such limitations into genuine humanistic understanding of how humans think. In disucssing "Acoustics versus Scholasticism", he writes:

> *The broad implications of what Sarton called "the cause and cure of scholasticism" may seem strange grist for the acoustical mill. There were at least two reasons, however, why the branch of acoustics dealing with music was able to make a unique contribution toward the ultimate conquest of scholasticism. The first was that music was assured a firm continuing hold on its place in the scholastic sun by virtue of its role as a part of the classical quadrivium....*
>
> *The second basis for the close relation between music and scholasticism stemmed from the fact that music is, sui generis, an epitome of experimental science. Objective in execution and humanistic in appreciation, its three aspects of composition, performance, and appreciation exemplified -- and held up continuously for conscious or unconscious regard -- the scientific credo of hypothesis, experiment, and conclusion.*

If as engineers we tend toward a bias for the objective, TED UZZLE found an understandable middle ground for this type of conflict when he said (about a different book, however) "I always thought of it in terms of the Kierkegaardian subject-object dichotomy, thus:

while you and i have lips and voices which
are for kissing and to sing with
who cares if some oneeyed son of a bitch
invents an instrument to measure spring with?
-e e cummings

Hunt's integrity of research, his patience over decades of data gathering, and his desire to express it all with clarity, understanding, and inspired scientific insight results in the reader being able to share the thoughts of his ancient predecessors rather than hearing about remote disconnected epics.

Delightful side trips include how Posidonius' miscalculation of the earth's circumsference later used by Ptolemy and then 14 centuries later turning up in the hand of Toscanelli to Canon Martin's to Christopher Columbus (the error made east to west distances much shorter than they were). This all led directly to Columbus' attempt to reach India by a Western route. Proof again that an error can and has led to substantial results. How often an error that results in some action being taken beats the most gifted intellect sitting on his hands.

We recommend this book most heartedly to those who love music and its production, reproduction, and reception. It's a thrill to encounter the earliest thoughts man has recorded on a subject dear to your heart.

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AES ANTHOLOGY ON SOUND REINFORCEMENT published by the Audio Engineering Society, $19.

The preface to this anthology on Sound Reinforcement credits Don Davis, me, with having helped select some of the papers. The papers that I selected for inclusion in the Anthology are: A-1, A-5, A-16, <u>A-23</u>, <u>A-28</u>, A-31, A-32 A-39, <u>B-1</u>, B-4, B-6, <u>B-13</u>, B-18, B-26, <u>B-51</u>, B-58, <u>B-68</u>, C-1, C-10, D-31, D-33, D-35, D-40, <u>D-41</u>, <u>D-59</u>, D-72, D-81, <u>E-23</u>, E-34, E-40, <u>E-46</u>, E-57, E-62, E-67, E-86, E-92, E-96, F-15, F-20, <u>G-1</u>, <u>G-9</u>, G-14, <u>G-25</u>, G-35.

The underlined numbers indicate Key papers possessing special information of unusual value.

What's fascinating in reading such a collection of papers is the directly contradictory statements made by authors of adjacent articles and on occasion by the author with regard to an early article by himself. The object lesson would seem to be "follow the author only so far as he follows what you can demonstrate to your own satisfaction as true."

A quick perusal of the section "Related Reading" at the end of the anthology reveals that the AES, while a worthy latecomer, simply was too late to be the Journal to receive the seminal papers of our industry.

The beauty of this Anthology is that it saves you the effort of having to individually collect all the papers I have noted above. If you were to remove the remainder this collection would be less likely to mislead the innocent.

Those of you fortunate enough to possess all of the "Related Reading" material have a splendid beginning of an audio engineering library. At $19.00 the <u>AES Sound Reinforcement Anthology</u> is a genuine bargain.

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Tab Books continues to pour out a steady stream of "How to" books. Among the latest is: *HOW TO DESIGN, BUILD, AND TEST SPEAKER SYSTEMS* by David B. Weems, $6.95. This is an excellent buy with 336 pages packed full of interesting tidbits the author has gathered over the years. Like most books of this type, the author is unaware of how a loudspeaker and a room are acoustically matched, doesn't understand the difference between directivity factor, Q, and coverage angles, C_L, and thinks sensitivity is efficiency. Power is rated in RMS watts (continuous average power is, I'm sure, what Mr. Weems means). And, the book contains no bibliography. He discusses Doppler distortion without even mentioning Paul Klipsch or Jim Moir, which is like discussing the Bible without mentioning God. Another, perhaps, more serious problem is that instead of selecting the best designs in each category, he democratically let's you figure out what might be best from a mass of design choices.

In spite of these minor problems, the book is full of excellent construction tips, has some enclosures made out of *drain tiling* that look interesting, and has a simplified version of Don Keele's calculator equation for vented box frequency response.

My impression of Mr. Weems, from reading the book, is that he is an energetic, enthusiastic, amateur loudspeaker enclosure builder who has kept up with contemporary literature to a commendable degree and has done an excellent job of making his reader aware of the approach a scientific designer should take. I'm pleased to have the book in my library for its virtues and it will serve as a reminder that the unique data we are gathering with TDS needs to be put in a book.

Books of Interest, cont.