SYNERGETIC AUDIO CONCEPTS

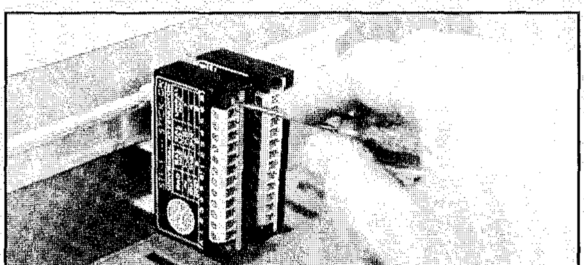
SYN AUD CON

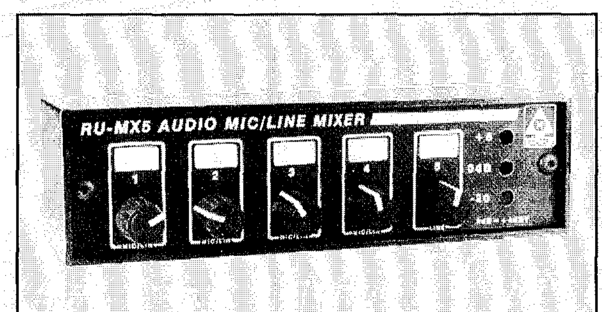
newsletter

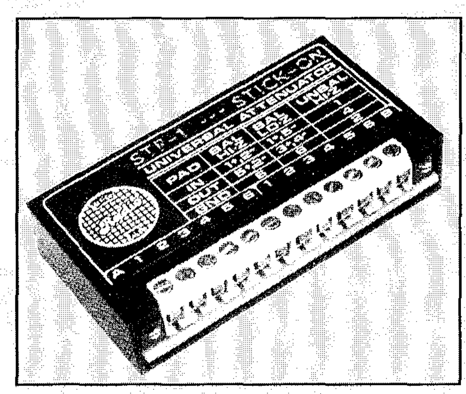
Volume 22, Number 3
Spring 1995

Pat Brown
Don & Carolyn Davis

A Contractor's Dream Come True...







Radio Design Labs

SYNERGETIC SYN AUD CON AUDIO CONCEPTS

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

Synergetic: Working together; cooperating, cooperative

Synergism: Cooperative action of discrete agencies such that the total effect is greater than the sum of the two effects taken independently.

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Special Supplement No. 1 to Newsletter Vol. 22, No. 3

No. 1 - The Business of the Sound Business

When do I renew? - You can check to see when your subscription will expire by checking the mailing label on the envelope in which your newsletter was mailed. In the upper righthand corner beside the name, a date will appear (i.e., 7-94). This means you will receive your last issue with that quarter's mailing unless you renew. Renewal notices will be sent one month prior to your last issue being mailed. You must renew before the next quarter's newsletter is mailed or your subscription will become inactive.

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About the Cover...

Syn-Aud-Con Welcomes New Sponsor

Radio Design Labs

Radio Design Labs was established in January of 1986 to provide precisely-engineered, practical products to a wide spectrum of audio users. Joel Bump, president of RDL, had previously spent years in the field encountering product needs, and experiencing firsthand the varying levels of customer support from different manufacturers. Joel teamed his design and systems engineering background with Jerry Clement's electronics sales experience to build the breadth of product offering and marketing outlets which make up Radio Design Labs today.

Fundamental to RDL is the philosophy of excellence, which is meticulously applied equally to product design, manufacture and customer support and satisfaction. Throughout the steady growth of the company, RDL has been fortunate to build a group of people who share the same basic concern for quality. All company operations were originally housed in Carpinteria, California in the building which is now used for sales and accounting. Manufacturing was relocated in 1987 to a more suitable Southern California location. In 1993, a transition was begun to move manufacturing to Prescott, Arizona, a permanent location with ample room to grow for many years. Just completed is a new office complex. The new facility has the capacity for several times the present production levels.

RDL's first broad-based product group was named "Stick-Ons" because of the mounting adhesive. The object of the Stick-On line was to provide compact, high-performance building blocks for system design. Most first-time users have been introduced to Stick-Ons as a solution for unique problems, often as simple as converting an amplifier line input to a mic input through the addition of a mic preamplifier. As the "building block" concept has grown so has the offering of Stick-Ons. Since the introduction of the line about 7 years ago, Stick-Ons not only offer a high quality solution to nearly any single audio problem, they offer the systems designer an unlimited array of functions which can be combined to provide the features required in the most demanding installations. The 40 Stick-On products offer every function, from audio preamplification, amplification, distribution, level control, limiting, audio detection and switching.

The need for compact high performance equipment which has connectors and user adjustments is met by RDL's Rack-Up product group. Rack-Ups can be used as stand-alone modules, or can rack mount three-across in a single rack unit, yielding not only unsurpassed performance, but also the highest racking density of any product line on the market. The highest quality potentiometers and connectors are fastened to steel subassemblies for structural integrity. Audio connections made "inside the rack" on the rear of the Rack-Ups are made through full-sized barrier blocks. Eleven products in the Rack-Up series offer a wide variety of audio and video functions, from preamplification to distribution and metering.

Both product lines are complemented by a wide variety of mounting options, making use of all the products simple and convenient. Individual mounting bracket kits are available for most products, and three rack-mount systems provide front and rear mounting of Rack-Ups, Stick-Ons, and combination mounting (various modules from each product line in a single rack unit chassis) and audio jack options.

Early 1995 has brought the introduction of 19 new products and models. Several new Rack-Up and Stick-On products are available (April 1st), as well as new mounting options and the new "TX" series of transformer-based interface modules.

One design philosophy remains constant. RDL provides the customer with the assurance of superior design and service. While some companies design products around hybrid "already designed" circuits from IC manufacturers, RDL designs are "ground-up" so the customer can be assured that the performance meets the industry's needs and the product group can be supported long-term by RDL. This "engineering-oriented" manufacturing approach has been well-supported by the industry.

Radio Design Labs is the first source many system designers turn to when "no compromise" performance is as important as compactness and economy, and for this reason we are proud to have RDL as a Syn-Aud-Con sponsor.

Winter 1995...

Until the first week in February, the winter of 94-95 has been a pleasantly mild one with above average temperatures. On February 4th, a classic piece of winter arrived heralded by "corn" snow driven by strong winds, followed by a good 4" snowfall and a temperature drop below zero. Once the temperature moderated we had two days of excellent skiing at our local ski resort (yes, we do have man-made snow with beautiful facilities - only drawback is it isn't much longer than 1/4 mile).

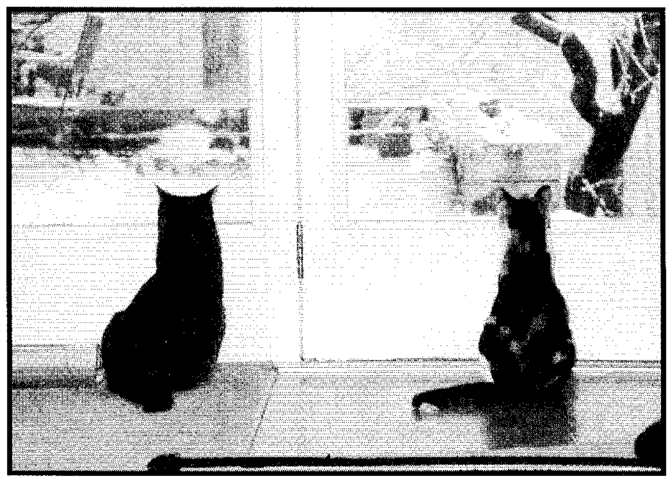
Carolyn and I have become addicted to a morning walk with the five dogs. On mornings after snowfall, trails we travel are encyclopedias of game animals - quail, grouse, mink, fox, coyote, muskrat, coon, squirrel and herds of deer. As I write this, Carolyn and I have just returned from a hike through the woods with the five dogs. Watching their antics, as Patch (the German Shepherd) goes into high gear after a rabbit, shortly followed by a full train in pursuit, is a major winter entertainment. As soon as all five dogs had vanished over the horizon, the rabbit, having circled back, nearly ran into Carolyn and I. He stopped for almost a minute, looked at us with care, and did a right turn to go around us. Patch, who can move through the woods with the silence of a ghost, soon returned, but my first warning she was back was her gently brushing my leg to notify me she was there.

Versus has delusions of management, rushing about smartly, barking actively, and charging all the returnees from the chase as if to say, "Report to me!" Both Pedro and Patch, as senior members, let Roe, Versus, and Wade do the leg work while they supervise at the center of the very large circle.

It's Wade these days that most often is the cause of deer moving into our view, if not into his. Wade caused a deer and a fawn to move into our view and three wild turkeys flew into the tops of large oaks in the woods.

Carolyn strides out ahead of me these days, and I'm sure if there were observers, they'd remark, "That's what he gets for marrying a younger woman." For a long time she was too busy to hike so I don't mind being the caboose to our parade.

We headed home, richer for our efforts. Back from the woods the warmth of the wood stove embraces the interior of our little house and leaves us with a sense of marvel that three split pieces of log can heat an entire house. I know of no more satisfactory form of artificial heat. The cats are lined up at the atrium windows looking at the hundreds of birds feeding on the deck.

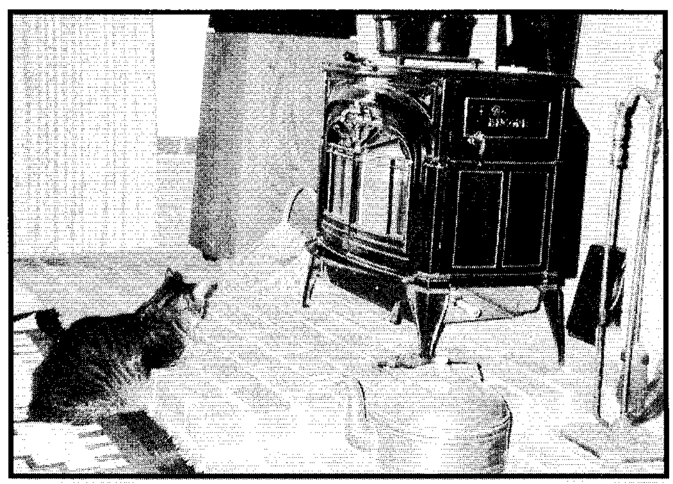


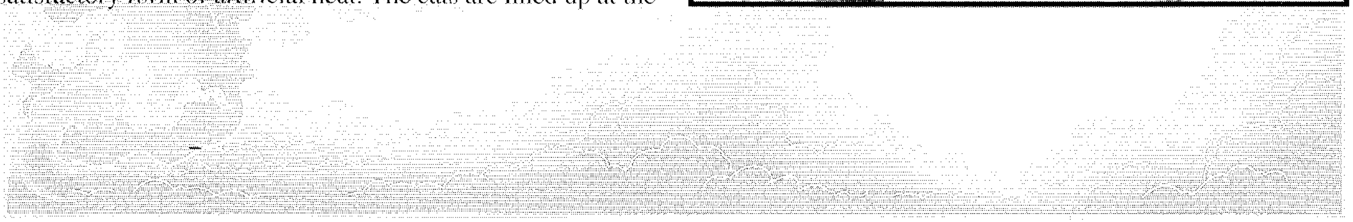
Our walk let's us see the comings and goings of the deer herds. I recently saw 15 deer in one group cross the front fields and go down into the woods by our spring.

An old rancher once told me that all that was materially necessary to man was a good meal, a warm shelter, and a place to lay his head. I would add to that a library of books and an abundance of music.

Not all is nature walks on a farm in the winter. Lots of wood to saw and split, snow to shovel, fires to be built and maintained. As the days lengthen we allow ourselves to think about Spring and getting ready for summer classes at the farm.

dbd





Up and Coming

SEMINARS AT FARM SEMINAR CENTER

The Farm - Norman, IN

Limited to 12 participants

May 17-19, 1995

June 8-10, 1995 New!

July 20-22, 1995

Sept. 14-16, 1995

October 12-14, 1995

3 - Day Seminars $550

SEMINARS ON-THE-ROAD

April 18-19, 1995 Tyson's Corner, VA

April 26-28 Indianapolis, IN (3rd day Dr. Wolfgang Ahnert)

Fall 1995, Boston Area

2 - Day Seminars $550

For Registration Information
(812) 923-0174 Ofc. (812) 923-3610 Fax

Richard C. Heyser and the TEF 20

Richard C. Heyser passed away eight years ago (March 1987). Those eight years have seen Dick's dreams for his measurement concepts develop into today's TEF20 analyzer with its truly remarkable software.

Jerry Stanley, Ron Bennett, Farrel Becker, supported and led through those eight years by Don Eger, produced the most complete audio and acoustic measurement system available today. Any mention of names is unfair to the myriad of other workers on the project but there is no denying the key roles the individuals mentioned played in achieving what we enjoy today.

The Sound Lab RTA option alone is a tool that eclipses any RTA we have used in the past, especially with its dual-channel capability.

Sound Lab PEQ, a joint development of Gene Patronis, Don Davis and Ron Bennett, is rapidly becoming the optimum way to equalize a sound system. For the very first time it is possible to apply precision equalizers with visible precision of measurement. Farrel Becker's minimum phase highlighting feature is invaluable in gaining an understanding of what can be and what shouldn't be equalized.

There are other worthy instruments on the market for various specialized tasks but nothing that has the breadth and scope of the TEF 20. While we like tools like the Ariel SYSid and the Audio Design units for electronic measurements, the TEF 20 does both electronic and acoustic measurements and is literally without a peer in the acoustical uses.

The Polar ETC program realizes a need of system and room designers that no one else is able to fulfill. As persons intimately involved in the development of the first 1/3-octave real time analyzers for the audio industry, and friends of Dick Heyser from the inception of his remarkable measurement insights, Carolyn and I feel that Dick would have been as pleased as we are with the present day manifestations of his genius.

Eight years have shown us that we are not likely to see his like again for another hundred years. Like all truly great men, his work has been picked at by petty intellectuals but his concepts keep right on being used by even his critics whenever accuracy and truth is needed in a measurement.

If you are in the audio industry, own a computer, and don't have a TEF 20, you are an incomplete entity. It's akin to having lived at the time of Gutenburg and chosen not to learn to read, but just to collect books. *dbd*

Richard C. Heyser 1931-1987

Upcoming Workshop...

The Theory of Design of High-Performance, Professional Loudspeakers

June 22-24, 1995

Staff:
Bruce Howze
Community Loudspeakers

Don Keele
Don Keele and Associates

Eugene Patronis, Ph.D.
Georgia Tech University

That mouthful of a workshop title tells it all— Real professionals who have real designs, successful in the market place, and who are willing to share what makes them both professional and successful. These are men who have designed to a standard that allows detailed measurement of their accomplishments without fear of coming in second best. Each member of this teaching staff has major accomplishments, genuine innovative leadership attached to their name including compression drivers, coaxial systems, constant directivity horns and many, many others. These men truly are their own peer group. They are fully capable of discussing High Performance Professional Loudspeakers: forming cones and diaphragms, adhesives, magnetic structures, enclosures, horn design, tooling, manufacturing techniques and advanced measurements.

With all of the above in mind, read over the subjects to be covered in the outline and ask yourself, "Isn't this an opportunity I shouldn't miss?" *dbd*

Dr. Wolfgang Ahnert to be Guest Instructor at Syn-Aud-Con Seminar

Wolfgang Ahnert, Prof. Dr. Ing., the co-developer of EASE and EARS with Dr. Feistel, is a good friend of Syn-Aud-Con and Don and Carolyn Davis. What this man has surmounted in his life is proof that when God's gift is given, not even being raised in the "Evil Empire" can destroy it. What most interests us about Dr. Ahnert, who we first met when the wall was still up and many of the worst manifestations of the regime that built it were in full force, is the fact that while they could restrict him physically he was a totally free man mentally. His joy in life, his gentleness, and his intellectual gifts made even his managers behind the curtain say to me, "He is our best."

With full freedom, Dr. Ahnert has rapidly received full recognition on both sides of the former wall. His remarkable mental energy, pent up from lack of adequate tools in the past, has burst forth in greatly advanced innovative, and highly useful acoustic analysis tools for the latest computers.

This day with Dr. Ahnert is a privilege you should not miss. You'll be blessed by meeting and getting to know a man with a vision of the future that we believe is truly accurate.

Attendance at this meeting is your chance to learn more about classic European thinking on room acoustics as well as the new perspectives on this classic material that TEF analysis has provided him.

While Dr. Ahnert is formatting the day for presentation of EASE and EARS plus what's ahead for each, it is his irrepressible nature and exuberant sharing that endears him to us. For anyone using or even thinking about computer aided design, April 28 is a vital date not to be missed. We'll look forward to seeing you there. *dbd*

Bernie Jakobs Retires From Shure Brothers

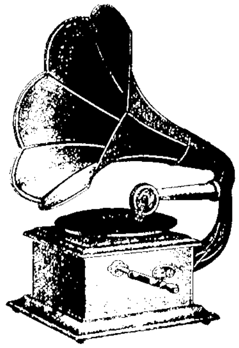
A 36 year career of continual growth with a successful company is an achievement few are likely to accomplish in our audio industry. We have known Bernie Jakobs for 23 of his 36 years at Shure Brothers. He started at Shure Brothers in 1958 as a design draftsman and ended his career at Shure as senior v.p. of engineering.
dbd

Bernie Jakobs

Out of the past...

HI-FI Defined

by A.C. Keller



Digging in my library I ran across the following sent to me from A.C. Keller, Murray Hill, NJ, in September 1980. It's a quote from H.A. Frederick, Rev. of Sci. Inst., May 1934.

Technical Requirements for Faithful Recording and Reproducing

(a) that the system have a linear relationship between input and output level,

(b) that the system have a uniform response vs. frequency characteristic since any complex wave may be resolved into the sum of simple sinusoidal terms, and

(c) that the system have a linear relationship between its phase shift and the frequency impressed upon it, and that this curve pass through $n\pi$ at zero frequency where n is any whole number"

That very succinctly describes the electronics chain of events. Acoustically we could add, for truly faithful reproduction:

(a) Spatial—directional—realism

(b) Reproduction of the Q vs frequency of original instruments, what happens in real life to performers and musical instruments.

Back in the days Keller is quoting, Bell Telephone labs tried to do:

(a) by developing three channel stereophonic sound and

(b) by having the Q of their horns shift gradually to a higher value at higher frequencies because that's what happens in real life.

All of the above still challenge today's serious designers.

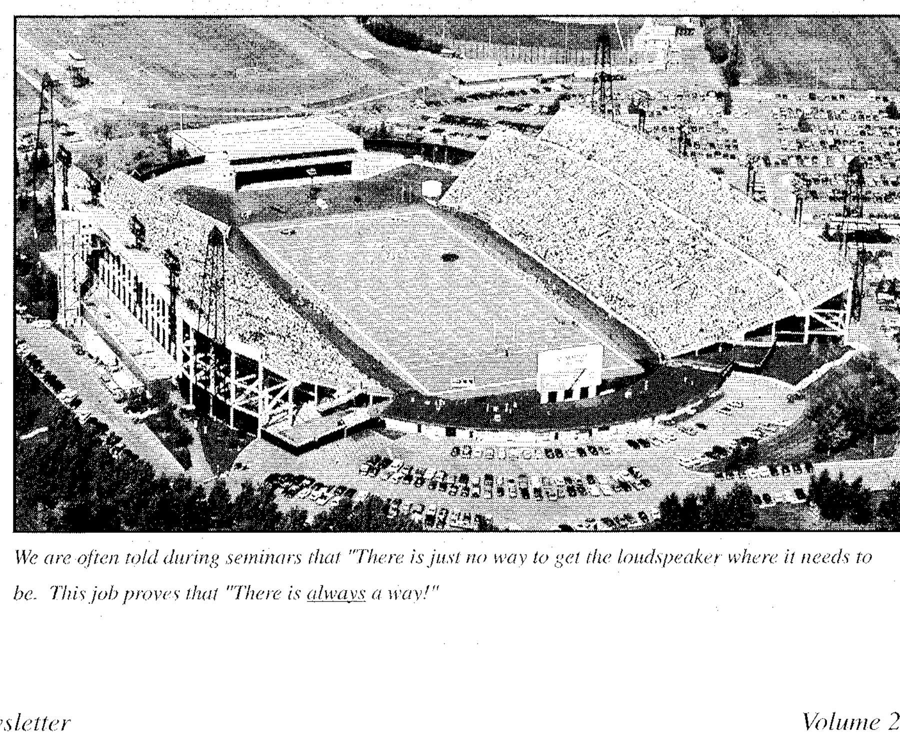
dbd

A Better View of "THE PROPER WAY TO DO A STADIUM"

In Fall issue of the Newsletter, Vol 22, N1, page 7, we published "The Proper Way to Do a Stadium." Several people called us to ask about the placement of the sound system in the picture. It wasn't perfectly clear that the sound system was hung over the center of the stadium. In going through the pictures to return to Dan Moran, who is the engineer who designed the sound system for Morgan, Dowhan Engineering, we found a picture that makes it very clear that the sound system in hung directly over the center of the field.

Note the 4 towers that hold the cables that supports the sound system. It is easy to see why the supports for the sound system cost as much as the components for the sound system. Structural engineers, Lamb McManus of Calgary, said that two-thirds of the 1.1 million sound system was for the structural steel and its installation by Alberta Government Telephone.

cpd



We are often told during seminars that "There is just no way to get the loudspeaker where it needs to be. This job proves that "There is always a way!"

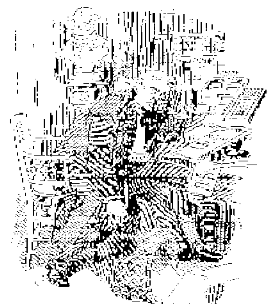
The Decision

In every human activity at some point you have to come face-to-face with either buckling down and learning the fundamentals or else faking it. To recognize reality, fakers can be financially successful and reap all the material rewards the same as the person working hard at learning the basics.

The reason to buckle down and go after the hard stuff has to do with how you feel about yourself. Do you want to know as much of the truth as is possible for you? Do you want to be perceived as intelligent or to actually be intelligent? Most men and women that I have been privileged to know who really had a firm grasp on fundamental issues have done so through an almost compulsive desire to understand what really was going on. They are also people who have learned to think.

In audio and acoustics the fundamentals are not difficult; the physics are.

That we "tread on forces" is self evident to the physicist, if not to the layman. That Ohms law, impedance, levels and other fundamentals work, even if not the absolute truth, but an excellent pragmatic approximation of something very complex, should not deter us from acquiring such tools.



The marvelous mystery of how energy manifests itself in our universe will never grow dull or uninteresting. *dbd*

Signal Processing

The words, "Signal Processing", are current hype words, so perhaps we should explore what can be processed and by what.

The audio signal, be it speech, music, or otherwise, has amplitude, phase, frequency content and temporal behavior as an electronic signal.

We can raise or lower the amplitude, (volume control); we can selectively raise or lower amplitude at selected frequencies (equalizer); we can delay it (signal delays); we can distort it (fuzz boxes, phasers, equalized crossovers); we can edit it (digitized rearrangement of context); or we can purify it (digital noise removal—noise, reverberation, hums, buzzes etc.). Today we can so process any audio or video signal that no rational person would be willing to accept at face value either audio or video testimony in a court of law.

We used to hear that, "Whoever has the most toys wins." But, today it's "Whoever has the most "bits" can win"

dbd

What Do You Think?

Quoting from a recent issue of EQ magazine,

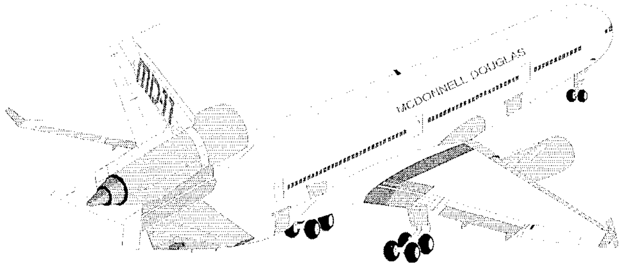
The digital domain EQ sounds good, and it does exactly what it says it is going to do. I like digital EQ the most when I have to brighten a vocal. If you have a vocal that tends to be a sibilant, then analog EQ only tends to exacerbate the situation. Digital EQ has less phase shift and therefore allows you to crank the high end...without making the esses tear your tweeters out by the roots.

Puristic comment: An accurately digitized audio signal implies that a high enough sampling rate has been used to allow the frequency response to extend to the upper limits of hearing (20 kHz), and that sufficient bits have been used to reduce to inaudible the quantization error in the amplitude of the signal (16-bit minimum). The goal of this process is to accurately capture an audio signal without coloration. If digital has its own "sound" then the process is breaking down somewhere. I solicit your opinions either pro or con.

E-Mail 74032,1356. *pb*

Man's business here is to know for the sake of living,
not to live for the sake of knowing.

Frederick Harrison



"The Airplane Guys"

Our "On the Road" classes always seem to bring us into contact with interesting people, and the Anaheim class was no exception. Mike Dills and Barney Graves from McDonnell Douglas were quickly dubbed "The Airplane Guys", and part of their duties include finding ways to increase speech intelligibility on commercial airplanes. This is a challenging task, since many of the solutions that the basic theories point to cannot be implemented for practical or logistical reasons.

On an airplane there is no reverberant sound field, and strong early reflections are minimal. For all practical purposes, the problem is an outdoor one, with inverse-square law and signal-to-noise the prime considerations. While this may at first seem simple to overcome, consider the noise levels on a typical aircraft. Noise levels typically exceed 80-90 dB "A" weighted, which means that the 25dB of signal-to-noise that we typically design for is out of the question. Ten dB is a more realistic figure, and many people will find Lp's of 90-100 dB objectionable. Having flown many airlines, most of which had extremely poor intelligibility, I have often pondered what I would do given the opportunity to attempt a correction of the situation. My "pecking" order would be as follows:

1. Loudspeaker placement - Move the loudspeakers into the seat backs, placing the listener's ears within a few feet of the sound source. Overhead loudspeakers do not take advantage of the built-in directivity of the human hearing system.

2. Gain Structure - According to Mike and Barney, there seems to be little standardization of the mics and mic techniques used by the pilot and crew. Noise cancelling mics and consistent micing distances would make optimum gain structure possible.

3. Announcement delays - While no one likes for their flight to be delayed, a digital delay on the announcements (6-8 seconds) would eliminate feedback, which the "Airplane Guys" tell us is a big problem on an airliner.

4. Prerecorded announcements - Digital recall of frequently used announcements would allow for a better and more consistent signal source for the system, making reproduction much more consistent.

5. Speech Processor - Speech processors like those available from Communications Company and Hughes can work seeming miracles when increasing level is not an option. The SP-1 from the Communications Company can give a subjective 10dB increase in signal-to-noise without requiring additional amplifier power or headroom.

It is our hope that "The Airplane Guys" will try some these things and report back to us the results.

We will keep you posted. *pb*

Just Barely Possible!

Fred Fredericks was the assistant instructor for the Anaheim "On-the-Road" class this past January. As we were installing a new motherboard into my computer, I was lamenting at why things always break when you need them the most. Fred summed it all up with one phrase, "Pat, you have to remember that all of this stuff is just barely possible!"

Well said, Fred! *pb*

Fred Fredericks

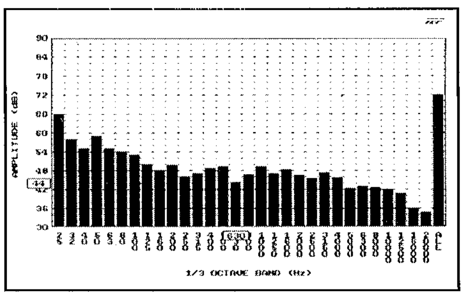
Real-Time Analyzers - Where do I put the mic?

RTA's are valuable tools for the systems designer/installer, but mic placement is critical if the data is to be believed.

Real-Time analyzers are valuable tools for system calibration. A good RTA serves to extend the capability of the technician, adding consistency and an additional vantage point to the trained ear of an experienced user. To achieve the greatest possible benefit from an RTA, proper mic placement is a must. Here are the factors to consider when answering the question, "Where do I put the mic?":

1. Critical Distance - To correct loudspeaker anomalies, the microphone should be in the direct sound field of the loudspeaker. Placing the mic too far from the loudspeaker results in a display that is heavily influenced by the reverberant sound field and/or early reflections. The equalizer that we are calibrating cannot affect these sound fields without making changes to the direct sound field. In properly implemented loudspeaker systems, the direct sound field should predominate at most listener positions. If it doesn't, consider using a higher Q source or shortening the throw distance. In most spaces, 10-12 feet from the loudspeaker is a good compromise for mic location. This is far enough away to avoid near field anomalies, yet usually well within critical distance.

In statistically reverberant spaces, an interesting exercise is to place the mic beyond critical distance to observe the L_W of the system. The L_W is the total acoustic power radiated into the space. Since this is what drives the reverberant sound field, taking a look beyond D_C can reveal octave bands that have insufficient or excessive acoustic power. This often reveals excessive bass energy, caused by achieving the desired bass level at a listener position with amplifier power rather than directivity. Such systems are typically plagued with low-frequency feedback underneath an overhead array of boxes.

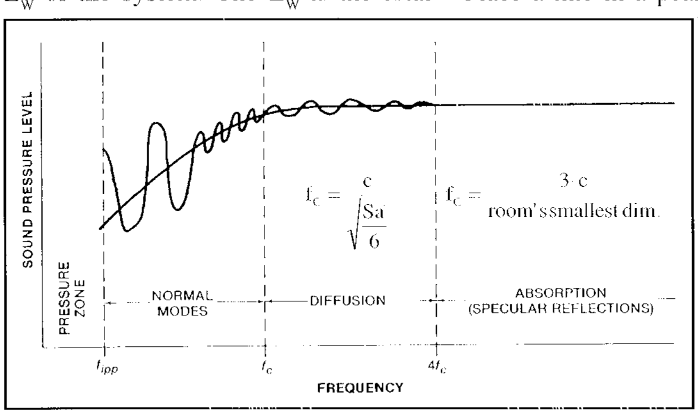


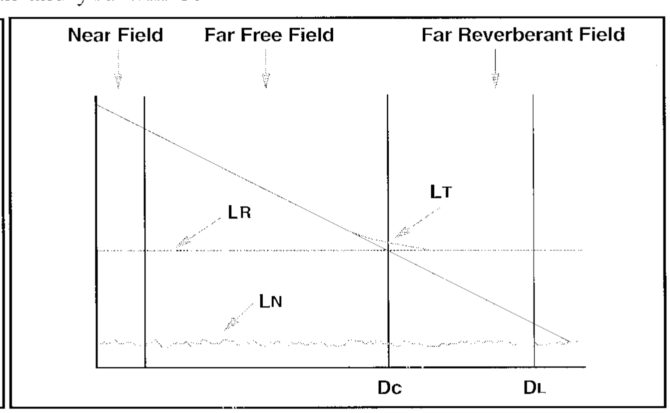
2. Critical Frequency - When the wavelengths emitted into an enclosed space are large enough (low enough in frequency), the room begins to selectively store these wavelengths longer than the high frequency (shorter wavelength) sounds. These are called standing or stationary waves, and are a function of the room's dimensions. When standing waves occur, the lower frequency bands on the RTA are purely a function (depend upon) mic placement. Place a mic in a peak and you will be profusely cutting frequencies. Place it in a null and you will be trying to "boost" the apparent missing energy. Since the equalizer cannot do two different things at two different locations, this exercise in futility should be avoided. The critical frequency of a space is that frequency above which the room acts like a large space, and below which it acts like a small space. This can be found empirically by roaming the room with an RTA and observing at what frequency the low frequency response begins to stabilize (100-400Hz in many spaces), or it can be calculated with one of the formulas in figure A.

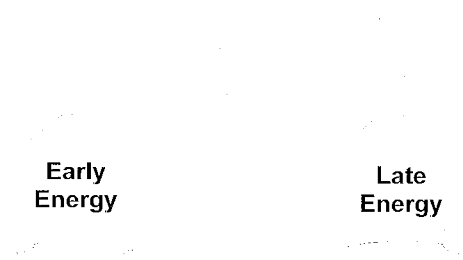
The response below critical frequency is something that is built-in to the room and hence something that you typically must live with.

3. Interference - Turn-on only one loudspeaker at a time - Comb filtering resulting from time (distance) spaced transducers cannot be corrected with an equalizer, for the same reason that room modes cannot: they will be different at every mic location. Since comb filtering of devices that overlap is something that we must live with, take care when aiming loudspeakers, to minimize overlap.

We hope that these simple guidelines will serve to streamline the equalization process and yield improved results. *pb*







A Balancing Act

Direct-to-Reverberant energy ratios are the key to optimizing a system for music or speech.

Cooking a fine meal involves selecting the proper ratios between ingredients. Sound system engineering is the same, only the ingredients have changed. Sound is produced in an enclosed space by radiating acoustic power into that space.

This acoustic power is experienced by the listener in a variety of ways. First, the original acoustic power produces a pressure wave that radiates through the space. When this wave arrives at a listener position, the resultant pressure variation at the eardrum is converted by the ear/brain system into the stimulus that we call sound. This small pressure variation represents only a fraction of the energy that was originally introduced into the space by the acoustic source, and is called the L_p, Sound Pressure-squared level (SPL).

As the pressure wave encounters the boundaries in the space (as propagation continues), the resultant reflections arrive at the listener position later in time, due to the increased travel distance. It is these "early reflections" that make one space sound different from another, giving the room an "acoustic signature." The spectral content and time distribution of the early reflections are a function of (depend upon) the room geometry and absorption present.

Our single sound source has now created many "virtual sources", as these reflections can be envisioned as individual loudspeakers distributed throughout the room. As the sound continues to propagate through the space, another sound field "builds up" as the energy accumulates. This is the reverberant sound field. Reverberation, which is quite different from early reflections, is a statistical accumulation of energy, and is limited only by the absorption in the room. In some rooms, the only significant source of absorption is the air itself, and reverberation times can be very long (many seconds). In truly reverberant spaces, this is the last part of the sound to decay to inaudibility when the acoustic source is silenced.

The sound of acoustical instruments is enhanced from the reflections and reverberation in a space. When a room is optimized acoustically for organ and choir music, amplified instruments seem to wreak havoc, and the results can be chaos. Dead spaces make good rooms for amplified instruments, yet organ and choral music sound thin and lifeless.

Rooms that are good for speech reproduction are designed to make use of early reflections (less than 35ms) which can be integrated with the direct sound by the ear/brain system, giving a perceived increase in the loudness of the sound. The effect of these early reflections is represented by the *early decay time* of the source/space combination. Since speech reproduction does not benefit from the later energy arrivals (greater than 35ms), we would like for the early reflections to decay as quickly as possible, yielding better definition to the direct sound field. A useful way of characterizing the music or speech quality of a space is shown below.

Note that the top of the chart is used for speech ratings (50ms split), and the bottom is used for music ratings (80ms split). In reverberant spaces, the sound system is used to change the E_D/E_R ratio at a given seating position, effectively moving the rating for that seat toward the right hand side of the chart. The question often becomes, "How do I increase the early energy at a seating position in order to achieve better speech reproduction?" First, to simply turn up the level does not change the ratio of early-to-late sound, since both sound fields are be-

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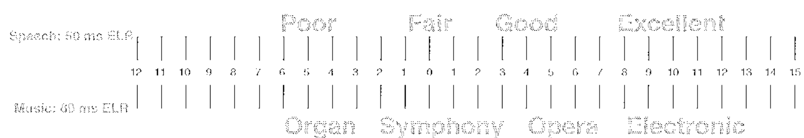
Early/Late Ratio Chart



Speech: 500 Hz, 1 kHz, 2 kHz, and 4 kHz. Intelligibility weighted and averaged

Music: 500 Hz, 1 kHz, 2 kHz, averaged

Courtesy Dave Klepper

continued from previous page

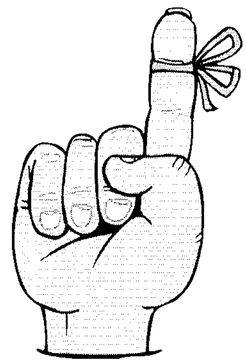
ing driven by the added energy. The early energy can be increased in one of two ways:

1. Shorten the distance from the listener to the sound source. The direct sound level increases by 6dB for each halving of distance to the source, so placing the listener closer to the loudspeaker will improve the speech quality in proportion to the distance moved.

2. Increase the directivity of the sound source. This results in greater intensity at the listening position, and can reduce the level of the reverberant sound field if the source is aimed at an absorptive audience.

Both methods are widely used, the former resulting in distributed loudspeaker systems due to the many loudspeakers required to get one close to everyone. The latter method is best implemented in carefully synchronized point-source loudspeaker clusters. These reduce the number of loudspeakers required, and minimize the acoustic energy input into the space, but require careful design to overcome the inherently longer throw distances. Each method has its advantages and pitfalls, which is why we need a system designer to weigh the choices against the customers needs.

Sound system design is largely a matter of reaching into the toolbox of basic theory and experience and arriving at the best solution for that particular application. *pb*



From the desk of...

Bill Beam

Image of the Imaginary

Bill Beam, Chief Engineer, WABC-TV, New York City, sent us the perfect message to have on an answering machine. He found it in the Lehigh University Alumni magazine:

"The number you have reached is imaginary. Please rotate your phone 90 degrees and try again."

Letters like Bill's are the fruitage we deeply appreciate.

dbd

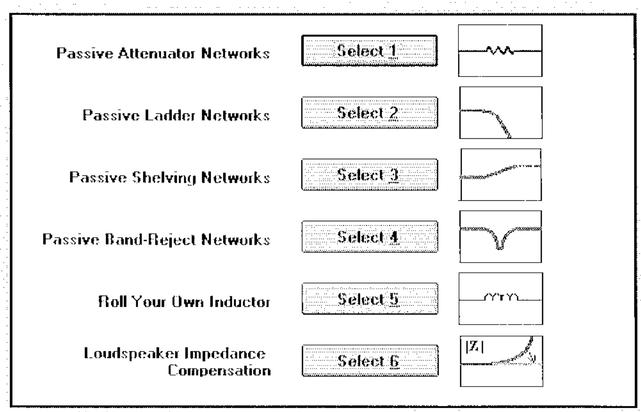
Filter Workshop

by Frank Ostrander

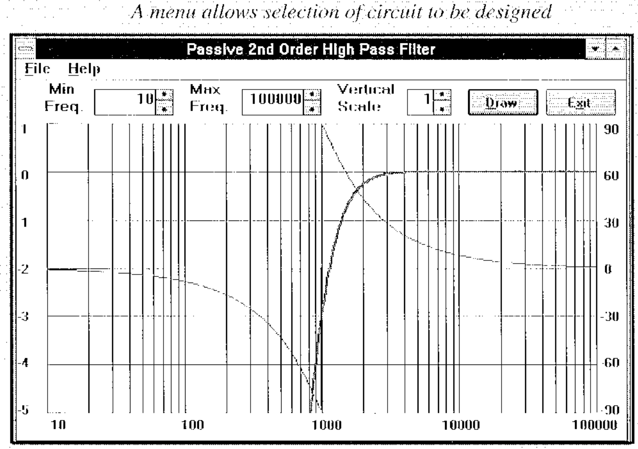
Frank Ostrander is a design engineer for Renkus-Heinz, Inc., a company which designs and builds high-performance loudspeaker systems.

Filter Workshop is a set of basic tools that are useful in the design of passive crossovers and other passive filter networks. It is intended for both engineering and educational use. As such, it attempts to provide enough background on the subject so that users can develop some insight for themselves. The program can assist in the design of passive crossovers, attenuators, inductors, ladder networks, etc.

Filter Workshop is a worthwhile program for anyone engaged in passive filter design. The $79.00 cost is a fraction of the cost of some similar programs. The Windows interface is straightforward and intuitive, and on-line help is available from most screens. Frank supports the program via E-mail, which allows the user to access his knowledge and experience should problems or questions arise.



A menu allows selection of circuit to be designed



Graphs display both amplitude and phase response

Technote

How to Measure Room Reverberation

Formulas for estimating speech intelligibility must have valid RT_{60} data to be useful. Here are some ways to get it.

We are often asked about available methods of measuring the reverberation time (RT_{60}) of a space. The RT_{60} is the time that it takes for the sound to decay to one-millionth of it's original level once a steady sound source has been silenced, i.e., 60 dB drop in sound power level. The threshold of hearing in humans is 20 micropascals, and the maximum pressure before pain is 200 Pascals. This is a 10,000,000 to 1 ratio (140 dB), from the formula:

$$20 \log \frac{200}{20(10^{-6})} = 140 \text{ dB}$$

Most public venues have a noise floor of at least 35 dBSPL, which means that our sound field must be at least 95 dBSPL for one to be able to hear 60 dB of decay prior to masking from the ambient noise. Keep this in mind if estimating RT60's "by ear." Analyzers do not need this much dynamic range, since they can observe the decay for a shorter span and extrapolate it for 60 dB of decay.

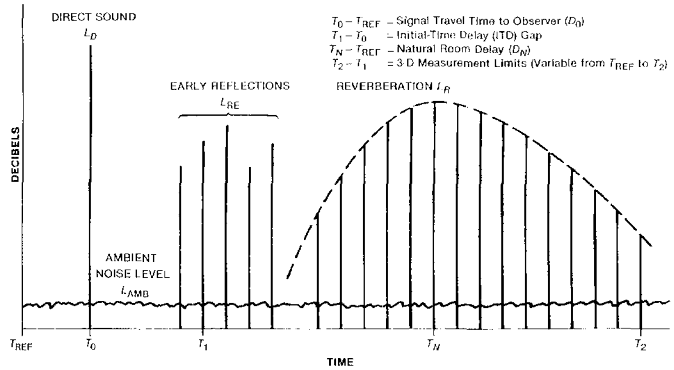
The longer that a reverberant sound field "hangs on" in a space, the more difficult it will be to provide intelligible speech. As one might expect, this is a frequency dependent phenomenon. RT_{60} is typically measured at each major octave band, with special emphasis on the 2kHz band.

Let's take a look as some of the methods:

1. Stopwatch - While some scoff at the simplicity of this approach, consider that it afforded Wallace Sabine resolutions into the millisecond realm. Two precautions are necessary. First, use band-limited noise as the source, so that each one-octave band can be measured individually (available on CD). Second, average the results to increase accuracy. I would be willing to bet that a room that consistently measures about 3 seconds for the 2kHz octave-band is pretty darn close to 3 seconds, certainly sufficient accuracy to plug into our intelligibility estimating equations.

2. Reverb Meters - Several companies offer meters that directly measure RT60. These work well when applied to spaces where such measurements are relevant (i.e. RT60's greater than 1.5 sec). The Goldline and Communications Company meters provide consistent and accurate results in the hands of the competent operator.

3. Computer-based analyzers - The advantages of this method are legion, and considered essential by many serious practitioners. An energy-time curve (ETC) shows the entire decay of energy in a space. Reverberation can be distinguished from early reflections, and the effects of each on speech intelligibility can be easily assessed. Since sound fields in rooms are extremely complex, to apply a one number rating (such as RT_{60}) can, in many cases, be a gross oversimplification. Rooms

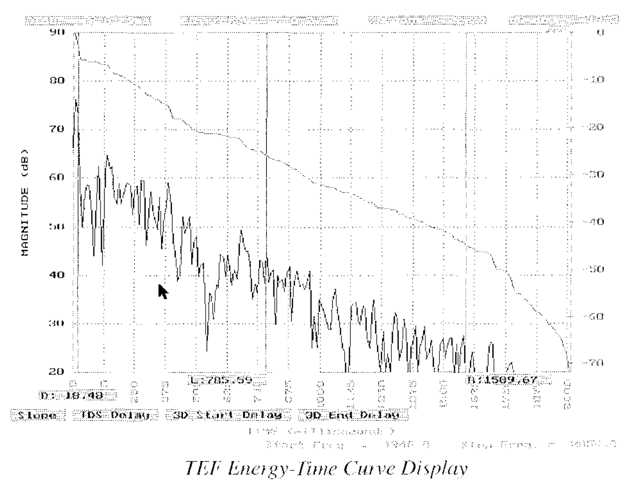


Sound fields vs. time in an enclosed space

with low RT_{60}'s can still be bad for speech if care is not taken to control early reflections.

In such spaces, (and there are many), our choices are quickly narrowed to the computer-controlled analyzers (TEF 20, Ariel SYSid) which are calibrated and consistent, or the ear/brain system, which is neither calibrated nor consistent.

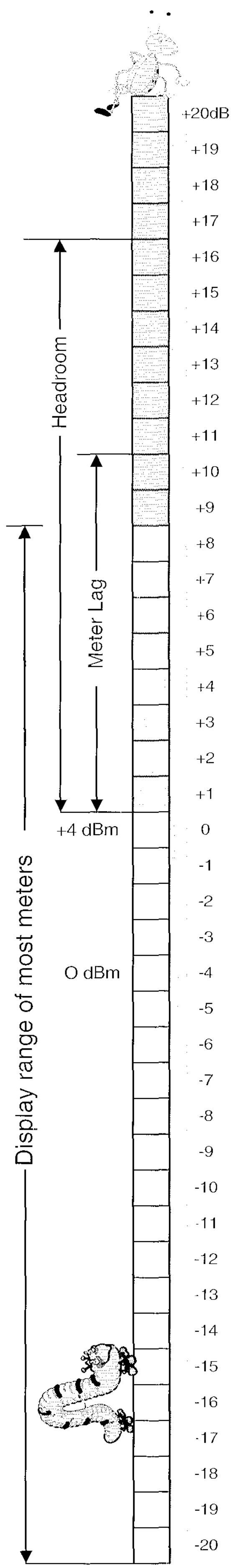
Rooms in which the reverberant sound field is the field of interest when considering intelligibility are usually limited to cathedrals, concert halls, and other diffuse spaces. These rooms have true statistical reverberation. Control rooms, class rooms, meeting spaces, etc. are most often dominated by the early reflected field, which calls for *specific* acoustical analysis and treatment rather than statistical. *pb*



TEF Energy-Time Curve Display

The VU

Getting the "bugs" out of your gain structure



The "volume indicating instrument," VI, is calibrated in steps of VU and percentage. The instrument does not read in either VU or percentage except as an incremental change. That is, where the indicating needle points does not tell the level in VU or the percentage power: a change in the indicating needles position from one position to another position does tell how many VU units the change represents.

Many assume that the VU stands for "volume unit" because the predecessor unit to the VU was the "transmission unit" TU,. Unfortunately, this is not defined in the standard or in any of the papers written by the members of the standards committee, and therefore, we use VU.

In a properly calibrated circuit, with an instrument that meets the standard, zero VU equals 0 dBm for a sinusoidal input. When such is the case, where should the indicating needle be pointed? The answer is at the -4VU mark. The reason for this is because the properly calibrated circuit with an instrument that meets the standard will have as its most sensitive attenuator setting, +4VU. The level, as stated above, is not read from the instrument but rather is the algebraic sum of the indication on the instrument plus the attenuator setting (-4) + (+4) = 0.

The original standard required the attenuator. Today many perfectly accurate volume indicating instruments are active FET devices of very high impedance and sensitivity. They are, however, internally adjusted so that a true sinusoidal level of zero dBm causes the needle to point at -4VU. This is logical since new and old instruments can be used side-by-side without confusion because they provide identical instrument readings.

The Purpose of Zero Indication

The intent of the standard was to have the instrument indicate, in normal operation, zero. Program material was to be adjusted via the attenuator until a zero reading was obtained and then the absolute or "true" level was read from the attenuator setting.

Ballistics

Anyone who has ever tried to "mix" program material by watching the level on a "peak" reading instrument knows that impulses that are not loud drive the needle to the peg. Human hearing does not detect short duration, high level peaks as loud. The hearing system integrates over short intervals to obtain an estimate of "loudness". The ballistics of the standard volume indicating instrument are adjusted so that it integrates the energy it is receiving over approximately 30 milliseconds.

In practice this results in the instrument indicating a level approximately ten decibels below the actual peak power. This is called instrument lag and it is accounted for by providing ten decibels more power capability in the mixing equipment than is being indicated by the instrument.

Headroom

Peak energy in program material fluctuates up and down and because of this, it is customary to provide "headroom" for these variations. Six decibels is the customary value.

The combination of the instrument lag and the head room requirements mean that to mix program material at a +4VU level, the mixer should have (+4) + (+10) + (+6) = +20 dBm power capability.

Looked at differently, this would mean that a mixer with a +18 dBm output capability should be operated at (+18) - (+10) - (+6) = +2VU. On an instrument that indicates zero for +4VU, then -2VU indication is the maximum program level you would want to see. *dbd*




Syn-Aud-Con Goes to Anaheim



January took the "On the Road" class to southern California, for a seminar that followed the Live Sound Workshop and preceded the NAMM Show. The attendees included sound people from all areas of the business, including manufacturing, industry, and the theatre. The basics of audio and acoustics are applicable to all of these areas, and each grad is left to apply them as needed. Fred Fredericks, former general manager of TOA Electronics, was guest instructor and offered valuable insights from his years of experience. We had a special treat on the first day, when Ken Wahrenbrock, original builder of the PZM microphone, and John Humble, our former Syn-Aud-Con rep, dropped in for a visit and lunch. We will be returning to the Anaheim area in June of 1996, and look forward to seeing old friends and meeting new ones. *pb*

Famous Last Words...

You can put it together yourself in five minutes.

That's not poison ivy.

I don't burn, I tan.

When it says empty, there's always a gallon or two left.

One piece of cake won't blow my diet.

Of course there's film in the camera.

Take off your clothes, the doctor will be with you in a minute.

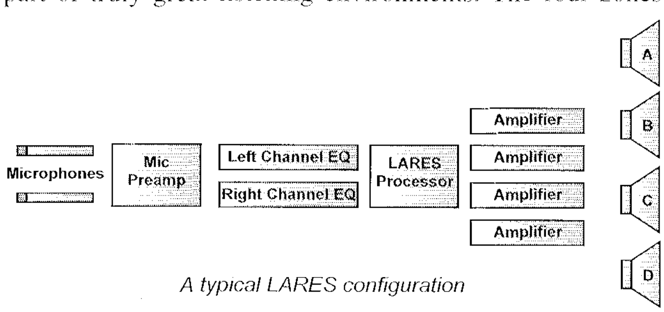
Creating Reverberation

Electronics can enhance an otherwise "dry" venue

Constructing a diffuse reverberant space is a difficult task. By diffuse, we mean that the surfaces in the space are small when compared to the wavelength of the sound wave. This means that large, flat surface areas must be avoided, because they do not "break up" the sound wave, resulting in discrete reflections. Excellent diffusor panels are available from companies such as RPG Diffusors, and this is often the best correction for a "flutterfull" room. Another alternative is the use of digital signal processing to provide a reverberant sound field where one doesn't exist naturally.

The LARES (Lexicon Acoustic Reinforcement and Enhancement) system represents the current state-of-the-art in such systems. The approach is straight-forward; utilize a couple of mics to pick-up the performer(s), process the signal with the DSP, and drive four distributed loudspeaker systems with the resultant reverb. While it sounds fairly simple, the actual implementation is quite complex, as several major obstacles had to be overcome by Lexicon's engineers. The first problem is feedback, since there are live mics and loudspeakers in the same acoustical space. This problem was solved by making the reverberant energy *time-variant*. LARES generates time-variant energy which simulates continuous movement of the loudspeaker. Before feedback can occur, the energy coming from each loudspeaker changes. This time-variant energy is reproduced through multiple audio channels. The result is remarkable freedom from feedback (typically 6 to 18 dB).

Another engineering obstacle is solved by utilizing multiple zones of loudspeakers. A true reverberant field has no directivity since sound waves impinge on a listener position from random directions. This "acoustic chaos" is an essential part of truly great listening environments. The four zones

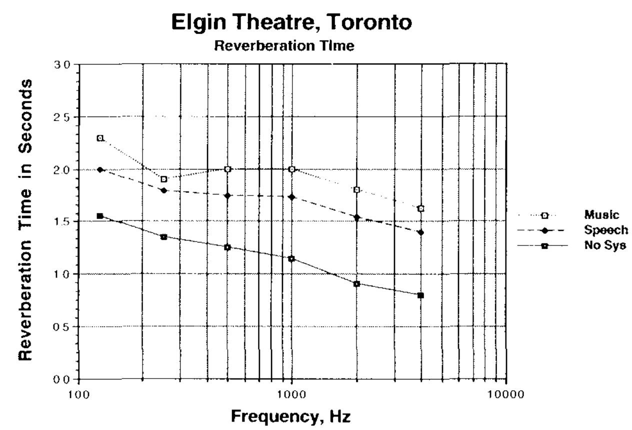


A typical LARES configuration

coupled with the time-variation of the energy provides a believable replica of true reverberation. Actual discrete reflections can also be added, allowing simulation of about any type of listening space.

Perhaps the greatest benefit of such systems is versatility. Imagine a hall that is used for opera on one night, choral music the next, a folk singer the next, and finally a high-energy rock group the next. With LARES, each of these groups can have the acoustics that best supports their music. A very dry church building can be constructed and the required acoustics added electronically. A 4 second RT_{60} can be selected for the opening organ and choir hymn, a 2 second RT_{60} for the soloist with sound track, and finally revert to the dry room for the sermon. Even then, a time-variant early reflected sound field can be used to reinforce speech (working within the Haas zone).

The cost of such is system is small compared to the architectural changes required to change the acoustics of a space.



Results of a LARES installation

Equipment prices from $40,000 to $80,000 are typical for moderate size concert halls and churches (350-2000 people). This places the price of a LARES system in the same category as a good pipe organ.

In new construction, LARES allows the architect and the acoustician greater freedom to focus on other important aspects of the space.

Since hearing is believing, we encourage interested parties to audition a LARES system. Lexicon can provide you with an installation nearest you. Syn-Aud-Con will be using electronic room enhancement during it's 1995 seminars to demonstrate and teach fundamentals of architectural acoustics and design of systems for difficult spaces, allowing us to get the data off of the charts and into your ears.

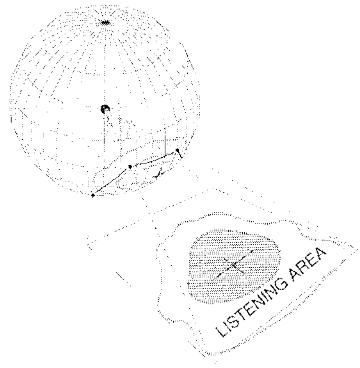
LARES systems have been installed all over the world, in venues such as the Elgin Theater in Toronto, Copley Symphony Hall in San Diego, and the Tsai Performance Center at Boston University, just to name a few. These venues have realized the importance of acoustics in their productions, and the LARES system has enhanced these auditoria far beyond the inherent acoustics provided by surface coverings.

We encourage forward-thinking consultants and contractors to investigate this growing and promising new market.

pb

PHD Plus™

New features enhance this valuable tool

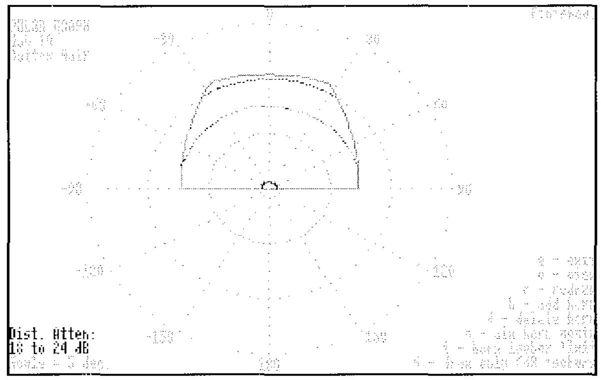


One point that is emphasized in Syn-Aud-Con seminars is that sound people should use and train their ear/brain processor, as it is the most valuable tool that we possess for doing sound system engineering. Used in conjunction with a scientific calculator or the Syn-Aud-Con slide rule, there are few system problems that cannot be quantified and solved. When the magnitude of the task exceeds the capabilities the basic tools, we turn to computers as a means of problem solving. A software tool that we consider extremely valuable is the PHD program from John Prohs. This excellent tool has been made even better with a recent upgrade of the program.

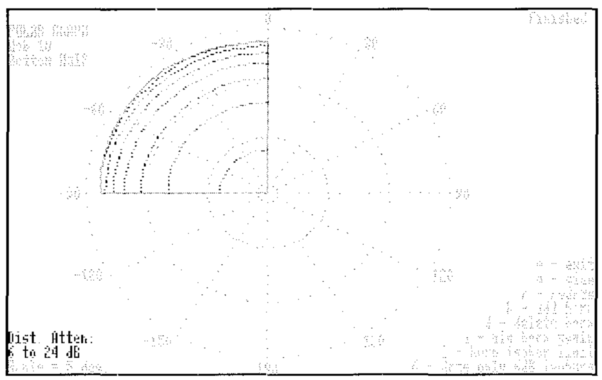
I have always liked PHD for it's simplicity in mapping in the seating areas. The system designer is then presented with an acoustical view of the room from the perspective of the loudspeaker. Inverse-square law level changes are indicated, as is the acoustic size of the surface to be covered. If the program stopped here, it would be a valuable tool. The ability to observe the seating areas from an acoustical perspective is an extremely useful aid in determining the best location for the loudspeaker. The designer can continue by trying components from the major loudspeaker manufacturers in the space. I was pleased to see isobars available for the Altec VIR and VIT series horns. The rectangular and trapezoidal patterns afforded by these devices are clearly shown by the isobars, allowing the designer to choose the optimum height for the device (see next article). Another valuable feature is the Q-Plus section, which allows the Q of any device in the database to be calculated. Q is an essential parameter for determining energy ratios for speech intelligibility in reverberant spaces, and one that many manufacturers do not supply data on.

PHD is available from John Prohs at a cost of $350.00.

pb



Seating area viewed from overhead. Note 6dB of inverse-square law level change from front to back.



Seating area viewed from side at stage level. Note 18dB of inverse-square law level change from front to back.

Some New Features of PHD+

* Greatly improved polar graph. The user can now toggle instantaneously between upper and lower views. Loudspeakers are now aimed by positioning a cursor, allowing you to think spatially and not just in terms of trigonometry.

* Performance analysis is totally different, and is now done graphically on the computer screen. The designer determines the desired window (eg: +/- 3dB) and then examines any desired seating position. If the point falls outside the window, it is indicated on the graphical display.

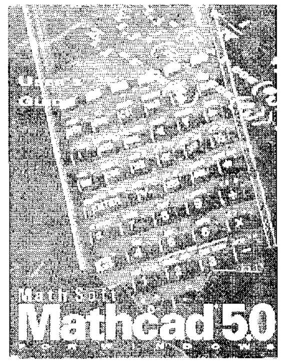
* Q Plus is now a part of PHD+. It can be called up from the main menu, making it no longer necessary to quit PHD+ to checkout horn patterns or to enter new horn pattern information.

* Icons have been included for use with Microsoft Windows.

* There are other amenities such as an easier way to change directories, better screen colors, more consistent menus, clearer instructions and presentation of data, and a much larger multibrand database of loudspeakers (over 2000 patterns representing over 500 loudspeakers).

These and many more improvements make PHD+ an excellent choice for the system designer.

Mathcad Quarterly *Playing with the numbers*



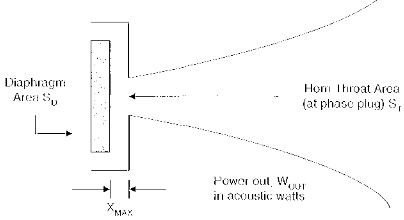
The topic for our first Mathcad Quarterly comes from John Wiggins at Community. John sent me a fascinating paper called "Sound System Design Using Mechanical Specifications of Drivers", written by Cliff Hendrickson during his tenure at Community. We have printed this in the past, but have never made it available on Mathcad so that you can play with the numbers. I found it quite fascinating, as it gets down to the basic ways that things work. Warning: consideration of the concepts described herein may change the way that you design sound systems!

For you Mathcad users, if you want the Mathcad Quarterly on disk, send $5.00 to cover shipping and handling. For you E-Mail users, put a note in my box and I'll upload it to you at no charge.

Send in your interesting Mathcad projects for future publications. *pb*

From Hendrickson's paper:

Sound systems, especially those used for musical sound reinforcement, are normally designed with respect to the bulk specifications of system parts. These are supplied by their respective manufacturers in the form of response, distortion, impedance, etc. The purpose of this paper is to provide an alternative look at system design, using predictable behavior of two simple loudspeaker mechanisms. These are volume displacement-limited low frequency peak power capability and "throat distortion." Using these, we can choose the number of components and number of frequency bands, including their crossover frequencies.



Go through the steps for each component in the system (low, mid, hi). Data is available from manufacturers, but here is a brief table:

Driver	f_{pg}	$S_D(in^2)$	$S_T(in^2)$	$X_{MAX}(in)$
TAD 2001	350	3.1	.39	.018
Altec 288	225	6.3	.63	.025
JBL 2440	225	12.5	1.25	.025
TAD 4001	225	12.5	1.25	.025
Community M4	150	40	8	.10

The conclusion? Work it out for yourself. (Hint: Three-way systems offer some very tangible benefits when low distortion is a design criteria.)

An Alternate Method of Sound System Design

a. Determine the RMS and peak levels of acoustic power required in watts.
b. Determine the maximum tolerable distortion vs frequency at the maximum power level.
c. Using this equation and the highest frequency of interest, determine the total throat area required from W and driver parameters.

$L_P := 105$ — One watt/one meter rating from manufacturer

$P := 100$ — Amplifier power in watts (average)

$Q := 10$ — Directivity factor of loudspeaker

$$L_W := (L_P - 10\cdot\log(Q)) + 20\cdot\log\left(\frac{1}{.282}\right) + 10\cdot\log(P)$$

$L_W = 125.995$ — **Calculated sound power level from loudspeaker in decibels**

$W_{outAVG} = 3.977$ — **Calculated acoustic power in watts**

$$W_{outAVG} := 10^{\frac{L_W}{10}}\cdot 10^{-12}$$

Equation 1.0

$f := 2000$ — Frequency of interest

$THD := 5$ — Percent 2nd harmonic distortion tolerable

$f_{pg} := 225$ — phase plug flare rate from manufacturer

$S_T := 1.25$ — Throat area from manufacturer

$$NS_T := 0.16\cdot\frac{W_{outAVG}}{S_T}\cdot\left(\frac{\frac{f}{f_{pg}}}{THD}\right)^2$$

$NS_T = 1.609$ — **Calculated total throat area required**

$$N := \frac{NS_T}{S_T}$$

Equation 1.1

$N = 1.287$ — **Calculated number of drivers required**

Make sure that the available throat area exceeds the throat area required. Use additional drivers if necessary. Next, determine the cutoff frequency of the driver.

$X_{max} := 0.025$ — Maximum excursion of driver from manufacturer

$HR := 16$ — Desired headroom

$$W_{outPEAK} := 10^{\frac{L_W + HR}{10}}\cdot 10^{-12}$$

$W_{outPEAK} = 158.308$ — **Calculated peak acoustic power in watts**

$N := 2.0$ — Enter a number greater than the calculated N from above.

$S_D := 12.5$ — Diaphragm area from manufacturer

$S_T := 8$ — Throat area from manufacturer

$$f_C := \frac{14.6}{X_{max}}\cdot\sqrt{\frac{W_{outPEAK}}{N\cdot S_D}\cdot\frac{S_D}{S_T}}$$

Equation 1.2

$f_C = 464.723$ — Cutoff frequency in Hz

Available horns may cause you to raise this suggested frequency.

$THD_2 = 1.773$ — **Calculated percentage of second harmonic distortion**

$$THD_2 := 0.4\cdot\frac{f}{f_{pg}}\cdot\sqrt{\frac{W_{outAVG}}{S_T\cdot N}}$$

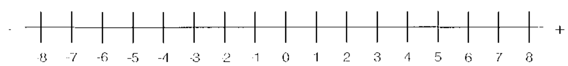
Equation 1.3

Mathematical Signs and Operators

Math basics provide insight into the basic principles of audio

It's little wonder to us that many who attend Syn-Aud-Con classes exhibit a fear of mathematics. What their fear really is, we have found, is fright of the way it's taught. When I went to high school and college before and during W.W.II, no one identified to me the difference between an operator and a sign, at least, not that I can remember.

Signs: + and - (plus and minus) are signs telling you what direction to go on what has been named an Argand Chart.



If I were to say +2, I would go two places to the right of zero (think of the plus and minus signs as if they were east and west indicators).

Operators: +, -, are also operators (as are x, and , log, sin, cos, and so on). Get out your calculator and try the following: Put in the numeral 2, press subtract (-) results = -2; press subtract again, results = +2. The subtract operator is also a change sign operator.

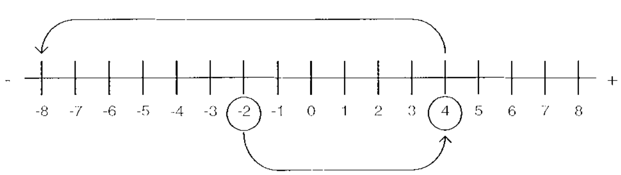
Example: (+4) - (-2) (+4) - (-2)

Signs Operator

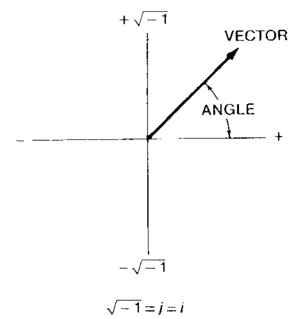
This on your calculator results in +6 why? Because starting at either -2 or at +4 there are six units between these two points. Another good example of the difference between signs and operators is

(-2)(-2)(-2)

(+4)(-2)



The times operator rotated the sign each time it was used with a negative sign resulting in a -8. A -4/-2 again illustrates rotation. Starting at -4 the divisor (operator) takes us to the positive side when its used with a minus sign. Starting at +4 a -2 divisor would rotate us back to the negative side. The -1, j, and i are operators that cause a 90 rotation rather than the usual 180 rotation.



If you'll look in *Sound System Engineering*, Second Edition, you'll find a series of examples on how these signs and operators are used in audio. Knowing mathematics from arithmetic through 'imaginary' numbers, logarithms and trigonometric functions are all usable on a daily basis. Higher mathematics are an eternal fascination and well worth individual study but certainly not something a competent audio person needs on a daily basis.

If the audio engineer hasn't reduced the calculus he knows to useful everyday algorithms, he's not likely to do so at a moment's notice. An audio engineer can scrape by without mathematics if he or she truly has an acquired knowledge of the basic physics involved. Unfortunately, that happy condition rarely happens short of thirty years of experience, plus remarkable starting ability on the part of the non-mathematical individual. *dbd*

Abstract ideas are the patterns two or more memories have in common. They are born whenever someone realizes that similarity...

Creative thinking may mean simply the realization that there's no particular virtue in doing things the way they always have been done.

Rudolf Flesch

Percent Alcons Clarification...

So far as I can tell, the latest TEF20 software calculates %Alcons from the Dir/Rev ratio using the same formulas as the TEF12 E-cursor routine. My question is: what is the formula?

The figures delivered by the TEF20 do not agree with the original Peutz charts or with the revised formula that Peutz presented at the 1987 Syn-Aud-Con intelligibility seminar. When I first noticed this I figured that a 'b' value of 4.0 or so had been included, but after running through a wider range of ratios, I can't find any consistent offset.

%Alcons calculations from some of the loudspeaker plotting programs don't match any of the above. Apparently everybody feels free to derive a proprietary modification of the Peutz formulas. However, if we are going to use %Alcons in sound system specifications and checkouts there has to be a standard way of deriving it.

I know the original E-cursor routine did not come out of thin air; it was based on input from Peutz and probably Don Davis as well. How is it calculating %Alcons?

Sincerely,

G. L. Augspurger

Carolyn Davis has forwarded to me your letter regarding the equations used in our TDS software for computing the %Alcons. I have enclosed a copy of these equations. They were provided in a letter to Gerald Stanley from Ben Kok (then an associate of Victor Peutz) dated January 10, 1985.

It is my understanding that the PHD program uses these same equations. I do not know about the others. I believe that most of them attempt to generate an impulse response via ray tracing - a touchy situation at best. They then compute the STI from this impulse response. If they present %Alcons, they probably derive it from STI but may be using the architectural equation if sufficient data is available.

I hope this answers your questions. Please let me know if I can be of further help.

Cordially,

Farrel M. Becker
Software Engineer

From Farrel Becker...

$$\mathrm{ISN} = -\log \frac{0.9}{\sqrt{1+10\frac{E_D}{E_R}}} + 0.1$$

where ISN = Information index due to signal to noise ratio

E_D = direct sound energy

E_R = reverberant sound energy

$$\mathrm{IT} = -\log \frac{0.9}{\sqrt{1+\frac{50^2}{RT_{60}}}} + 0.1$$

where IT = information index due to the reverberation time

RT_{60} = reverberation time

$$\mathrm{IPARR} = (\mathrm{IT}+\mathrm{ISN}) - (\mathrm{IT}\cdot\mathrm{ISN})$$

where IPARR = information index for parallel paths

$$\%\mathrm{Alcons} = 100\cdot 10^{-1.7\cdot \mathrm{IPARR}}$$

Author's Note:

The E_D and E_R values in the ISN equation must be converted into acoustic power (watts) using the following equation:

$$E' = 10^{\frac{E_x}{10}} \cdot 10^{-12}$$

The original equation that Mr. Augspurger was referring to is as follows (from *Sound System Engineering*):

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

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

$$C = -0.5\log\left(\frac{T_{60}}{12}\right)$$

$$\%\,Alcons = 100\left(10^{2[(A+BC)-ABC]}\right) + 0.015$$

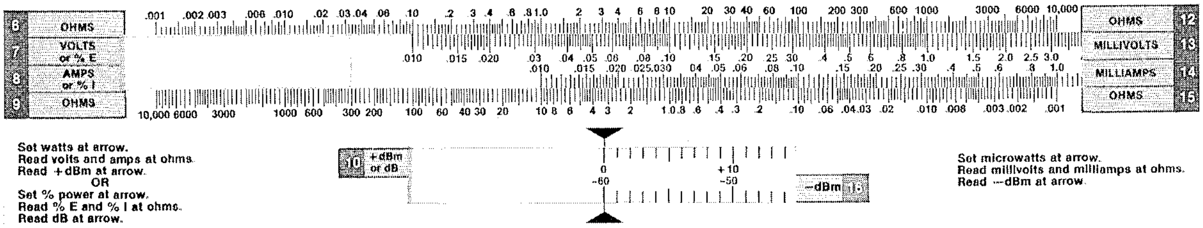
I created a Mathcad template to solve both equations simultaneously using E_D, E_R, and RT_{60}. The greatest deviation between solutions was less than 2% when well beyond D_C (E_R>>E_D) and less than 1% up close.

Since the TDS TEF %Alcons measurement does not account for ambient noise, one can take the E_D, E_R, and RT_{60} from the TEF measurement and place them into the above equation to estimate the effect of increased noise in the environment. *pb*

The Syn-Aud-Con Slide Rule

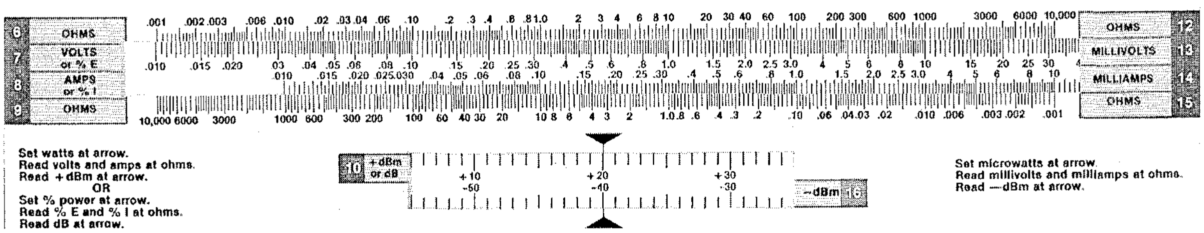
In coming issues we will be investigating some of the uses of this valuable design tool. Let us take a look at how the power equation section can be used for converting voltages into dBV, dBu and dBm values.

Here we go. On Scale 10, place 0 dB under the marker as shown below. We can now observe the dB reference values on Scales 5 and 6. Note that on Scale 7, 1 volt lies directly under 1000 Ohms. We will therefore use 1000 as the marker for converting voltages to dBV. Also note that 0.775 volts lies directly under 600 Ohms. We will therefore use 600 Ohms as the marker for converting voltages into dBu. As a result, any voltage lined up under the 1000 Ohms marker will read in dBV on Scale 10, and any voltage lined up under the 600 Ohms marker will read in dBu on Scale 10.

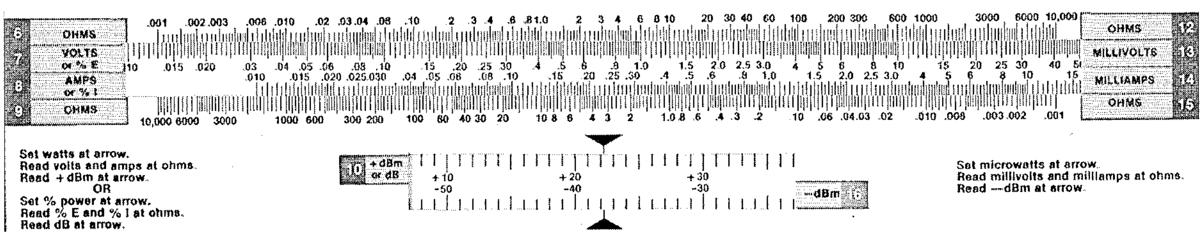


An exercise is in order. Let us consider a mixer whose maximum output voltage with a sine wave input is measured at 10 volts on an AC voltmeter. What is this value in dBV, dBu and dBm?

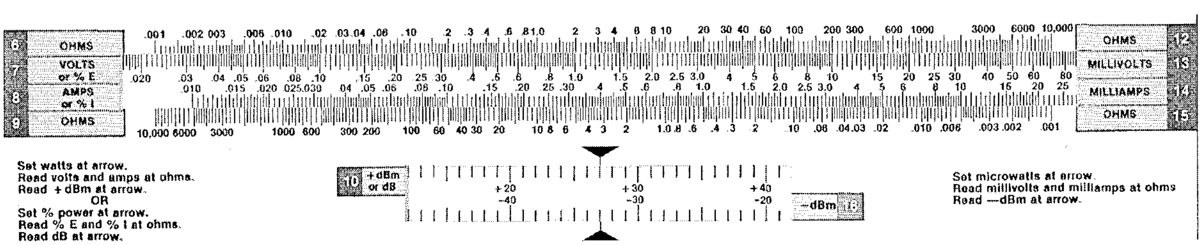
To find dBV, simply align 10 volts on Scale 7 with 1000 Ohms on Scale 6. Read +20 dBV on Scale 10.



To find dBu, align 10 volts on Scale 7 with 600 Ohms on Scale 6. Read +22.2 dBu on Scale 10. It is interesting to note that a dBu value is *always* 2.2 dB *higher* than a dBV value. (Maybe this is why people insist on using it!)



Finally, let us find the output level of our mixer in dBm. First, we need the output impedance of the mixer, since we are calculating power, and power concerns both voltage and impedance (Ohm's Law). From the manufacturer's spec sheet we determine this to be 200 Ohms, a typical value. Placing our 10 volts under 200 Ohms, we read +27 dBm on Scale 10. Since this is based on an open-circuit voltage, and only half of 10 volts would be available in the matched case, we subtract 6dB from our answer for +21 dBm.



I believe it is safe to say that you will find no better way to compute the gain structure of a sound system then this handy slide rule. It is available from Syn-Aud-Con for under $10, and doesn't require batteries! *pb*

Airmen of Note

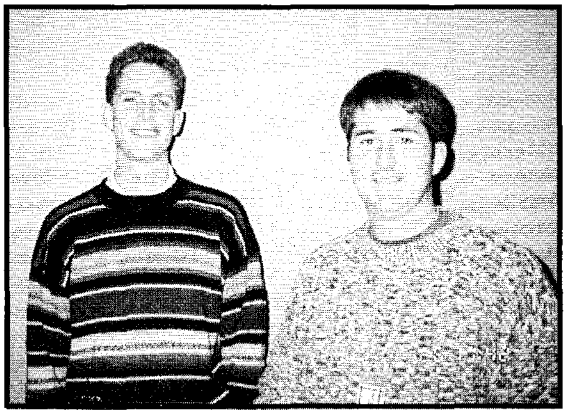
The United States AIR FORCE BAND
Washington, D.C

Ask most people to name the largest touring sound companies in the world and you will get names like Showco and Clair Brothers. Ask them to name the largest agents in the world and you will get a different list of names. All things considered, the military is probably the largest single organization that does all of these things, and a recent trip to Washington DC confirmed this in my mind.

The U.S. Air Force Band recently held it's annual training workshop in our nation's capital. Karl Winkler and Joe Dougherty played key roles in the planning and execution of the seminar. Seminar attendees included men and women from all parts of the globe the vast majority of which were musicians, a trend that we notice in most of our seminars. The military does an impressive job in outfitting it's musical groups with professional equipment, with extensive use of equipment from JBL, Soundcraft, AKG, Sennheiser and many others.

Karl Winkler explains that while the military brings in many nationally known acts for concerts, the vast majority of it's groups are from within it's own ranks. At Bolling Air Force Base alone, there are jazz groups, choral groups, a country band, a rock band, and others. These groups tour nationally and internationally, and are fully equipped with equipment and technical personnel. One of the most popular groups at the base are the "Airmen of Note." This is a top notch jazz band with several CD's to their credit.

It is hard to imagine a better place for young people with an interest in audio to gain knowledge and experience. Employers may find it difficult to lure the military technical personnel away from their positions that include good pay, excellent benefits, and a retirement plan. If this type of life interests you, stop in and see Joe and Karl at the Air Force Band booth at the next AES Convention. *pb*



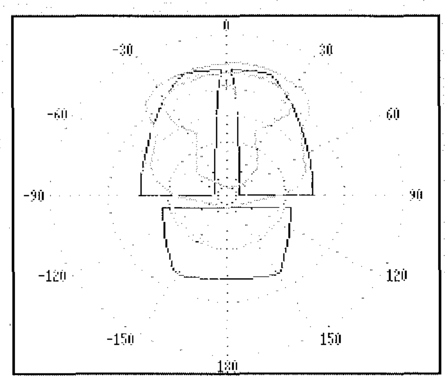
Karl Winkler and Joe Dougherty

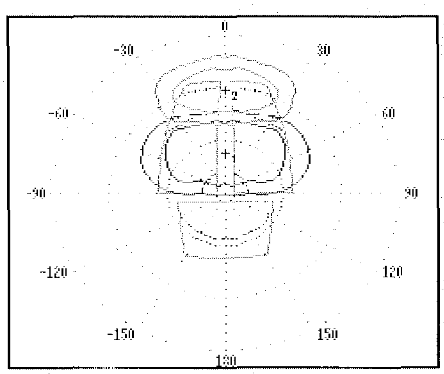
<u>*From Altec Lansing*</u>

Variable Intensity Horns:
Sometimes Just the Right Tool for the Job

We have seen many advancements in horn and driver technology in recent years. Synchronized coaxial horns have taken an important place in the system designer's toolbox, as have dedicated mid-range drivers and asymmetrical horns. Altec Lansing has provided the system designer with yet another valuable tool. The VIR (Variable intensity) horn has many applications in modern structures. A rectangular seating area can be covered by a single device for the mid and high frequencies, greatly simplifying design and reducing cost. The area of coverage of these large horns is dependent only on it's height, and Altec provides a slide rule to easily calculate this parameter. The VIT is a similar device, yielding a trapezoidal pattern rather than rectangular. Data for both the VIR and VIT are included with the most recent update of the PHD design program from John Prohs. Figure 1 shows a rectangular seating area covered with a conventional long-throw/short-throw design. Figure 2 shows the same seating area using a VIR.

We urge system designers to consider these devices when coverage requirements so dictate. *pb*





Surfin' the Net

Response to our invitation to E-mail users has been overwhelming, to say the least. We had no idea that there were so many of you "net surfers" out there. The potential for this type of communication is staggering, and we will continue to promote it as a way for audio people to stay in touch and exchange ideas.

E-mail is often simply a more practical way to communicate.

Top Ten Reasons for Using E-Mail

1. Little or no cost in sending and receiving messages.
2. NO PAPER!!! My desktop is cluttered enough.
3. Send your mail at any time, get your mail at any time.
4. Reply to messages with a click of the mouse.
5. NO PAPER!!!
6. Send about any type of file to about anywhere.
7. Let your computer send the same document to a whole list of people while you eat potatoe chips (Dan Quayle spelling).
8. Get neat stuff like the Mathcad Monthly without waiting for the mailman (excuse me, mail person).
9. NO PAPER!!!
10. Simply the best way to communicate with people overseas. (Have you ever tried to fax something abroad? It doesn't always work, regardless of what the commericals say.)

How Does it Work?

1. Get a computer and a modem (a fast modem, 14400 bps or better.)
2. Subscribe to an on-line service (Compuserve, Prodigy, America On-Line, etc.). Fees are usually about $10 per month. Access to the service is almost always a local phone call. Alternately, get hooked-up on the Internet. This costs more, but the possibilities are staggering. You can get Internet kits from any software supplier. Most on-line services have a "gateway" to the Internet, so that you can still send and receive docs to Internet folks without actually subscribing.
3. Once on-line, you have an address and "mailbox" that is unique to you. Simply check it each day for E-mail. You may also send documents to anyone else's mailbox, as long as you know their address.

Once on-line, there are many services that you may use free of charge as part of your membership, and many extended services that charge a small fee.

Syn-Aud-Con now sends out regular audio news updates as a part of your newsletter subscription fee. If you would like to receive the updates, just get us your address. Also, if you would like some information included with the next update, just E-mail it to me. Keep it short, simple and informative.

Get "on-line" today, and save time, long distance charges, and that money spent on curly fax paper. Stay better informed and develop friendships with people the world over. E-Mail is here to stay. *pb*

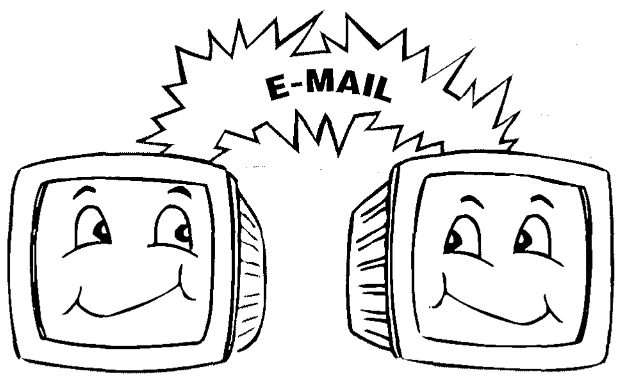
R. U. ON-LINE?

Syn-Aud-Con can now be reached via Compuserve and the Internet. One of our goals for 1995 is to enhance the interaction between the Syn-Aud-Con network of grads by promoting E-mail as a way of keeping in touch. This will allow grads to exchange information more efficiently and inexpensively than ever before. We can be reached at these addresses:

Compuserve 74032,1356

Internet 74032.1356@compuserve.com

If you are interested in receiving audio news, information, etc. via E-mail on a regular basis, please contact Pat Brown at 812-923-0174 or leave your name and address in our mailbox. *pb*



"Look It Up in Harry's Book!"

A recent Syn-Aud-Con Newsletter informed you about a book from the 1940's that we still find a valuable resource for audio and acoustical information. The book is "Acoustical Engineering" by Harry Olson. The subject surfaced again during a dinner conversation at the Dallas AES meeting with John Murray. John spent time at Electro-Voice during his career, a company known for their inventions and innovations. John said that when someone thought that they had come up with something new, they were told to go and "Look it up in Harry's book!" Don has always said that people spend too much time trying to "rediscover" things that were unearthed long ago, and that it is important to go back to the beginnings when trying to understand why things are so. Following is an excerpt from the book:

Harry Olson

In acoustics the ranges of intensities, pressures, etc., are so large that it is convenient to use a scale of smaller numbers termed decibels. The abbreviation db is used for the term decibel. The bel is the fundamental division of a logarithmic scale for expressing the ratio of two amounts of power, the number of bels denoting such a ratio being the logarithm to the base ten of this ratio. The decibel is one tenth of a bel. For example, with p1 and p2 designating two amounts of power and n the number of decibels denoting their ratio:

$$n = 10\log\frac{p1}{p2} \text{ decibels}$$

When the conditions are such that ratios of currents or ratios of voltages (or the analogous quantities such as pressures, volume currents, forces, and particle velocities) are the square roots of the corresponding power ratio, the number of decibels by which the corresponding powers differ is expressed by the following formulas:

$$n = 20\log\frac{i_1}{i_2} \text{ decibels}$$

$$n = 20\log\frac{e_1}{e_2} \text{ decibels}$$

One wonders why there is still so much confusion about things that were established long ago.

The book is still available, the current publisher being Professional Audio Journals, Inc. PO Box 31738 Philadelphia, PA19147-7718. *pb*

Makin' Copeeez!

When you call Syn-Aud-Con's Greenville office, where the day-to-day operation of Syn-Aud-Con takes place, a voice often heard belongs to this young lady. Brenda Brown, Pat's wife, spends alot of time in the office scheduling, managing, and making sure that Pat is working. Brenda can answer many of your questions concerning seminar details, upcoming workshops, and what that pain in your back may be. (Her "other" job is as a registered emergency room nurse.) *pb*

New for the TEF 20

Polar Energy-Time Curves

Techron continues to broaden the scope of the TEF 20 analyzer with the release of a major software package; Polar ETC. While we have all benefited from the information provided by Energy-Time Curves, Polar ETC goes a step beyond by providing directional information for the reflections arriving at a given microphone location. This is accomplished by using a directional microphone for the measurement. The mic is aimed a different direction for each measurement resulting in six ETC's (Energy-Time Curves). This information is post-processed into a polar view of the mic position (see figure). The magnitude and direction of each reflection is instantly displayed, allowing the operator to determine if the magnitude and time behavior of the arrival would enhance or detract from the listening experience. The density of energy arrivals from each direction provides valuable clues as to the overall behavior of the space for speech and/or music.

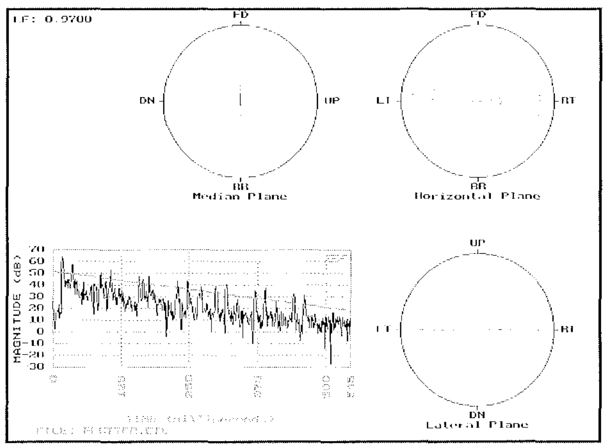
This is a truly three dimensional view of the reflected sound field as perceived by a given listening position. The menus, displays, cursor, and data presentations are precise and elegant.

So, what's it used for? right now two major uses suggest themselves.

1. Place a microphone via the PoGO laser exactly on the center line between two horns adjust alignment for the lowest possible reflected energy return.
2. Because it is easy to separate direct sound level from reflected sound level, it is possible to see the directivity advantage of one device over another.

Polar ETC's are a relatively new development in the world of audio measurements, being first implemented by Farrel Becker on the earlier TEF 10 and 12 analyzers. However, a look at history reveals the basic theory was used in World War II for sonar, and the "Hell's Bells" will live in infamy as a technological development that changed the course of history. As Don has often said, "The ancients keep stealing our inventions!"

Soundlab is a collective term used to describe the software group for the TEF 20 analyzer, and Polar ETC is a worthy addition to this family of software products. Those who have become accustomed to the benefits of directional energy arrival data would not consider doing serious sound work without it. As Deward Timothy, a talented consultant/installer from Salt Lake City put it, "I can finally retire my TEF 12!" *pb*



Polar Energy-Time Curve display

ACOUSTIC PARAMETERS

It is always wisdom to sit down once a year and review what parameters are known and available to us in our acoustic measurement work.

1. Acoustic level
2. Acoustic phase (both relative and absolute)
3. Acoustic power
4. Directional characteristics
5. Acoustic distortion
6. Acoustic temporal behavior
7. Acoustic impedance

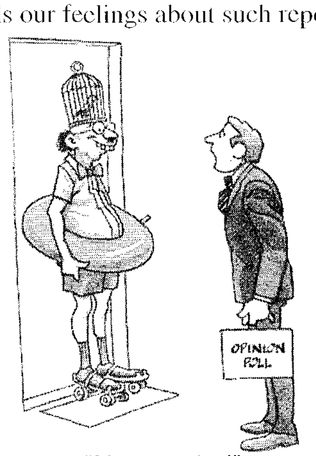
Some measurements combine several parameters at once. For example, the Polar ETC combines:

1. Acoustic level
2. Acoustic delay interval—distance
3. Acoustic directional characteristics (angular coordinates from observation point to acoustic origin)

dbd

REVOLUTIONARY PRODUCTS

We are sometimes asked why we don't report on the latest magic automatic signal processor or the metaphysical wire company's newest product. The cartoon shown here clearly reveals our feelings about such reports. *dbd*



"Never mind"

You Know You're In Trouble When...

Church sound is a challenging task, one and which you can do everything right and still "take it on the chin." My experiences with church sound bring to mind some situations that you make want to watch out for...

You know you're in trouble when...

1. You arrive to begin the installation and the youth group's there to help.

2. The minister has his own Radio Shack sound level meter.

3. The head of the sound committee is a ham radio operator.

4. Everyone is over 80.

5. The most often asked question is "What will it look like?"

6. The newest thing in the building is older than you are.

7. They say "No one's ever been down there."

8. The committee chuckles when you ask to see the building plans.

9. You raise the attic access panel and a can of "Yard Guard" falls on your head.

10. Your biggest competitor gave them your card.

pb

J.W. Davis Company

During the AES Conference in Dallas, I had the opportunity to visit with long time Syn-Aud-Con friend and sponsor, the J.W. Davis Company. J.W. Davis is one of the few companies in the business that is a complete supplier of audio products, capable of providing everything for the contractor; from loudspeakers, to amplifiers, to mic jacks. Few would argue that their wall-mount speaker baffles are among the best available anywhere. Having built a few loudspeaker enclosures myself, I can truly appreciate the detail that goes into their loudspeakers.

In addition to wall-mount baffles, J.W. Davis also manufactures the highly respected Pataxial loudspeakers, named after their designer, Dr. Eugene Patronis. This is one of the best two-way systems available, combining coaxial design with pattern control throughout most of it's bandpass. Company owner Harvey Earp explains that consistency has been their key to success, and over 60 years in the business proves that his formula works. A relationship with the J.W. Davis Company will enhance your ability to meet your customer's needs, and we recommend their services with confidence. *pb*

Harvey Earp and Jack Tucker

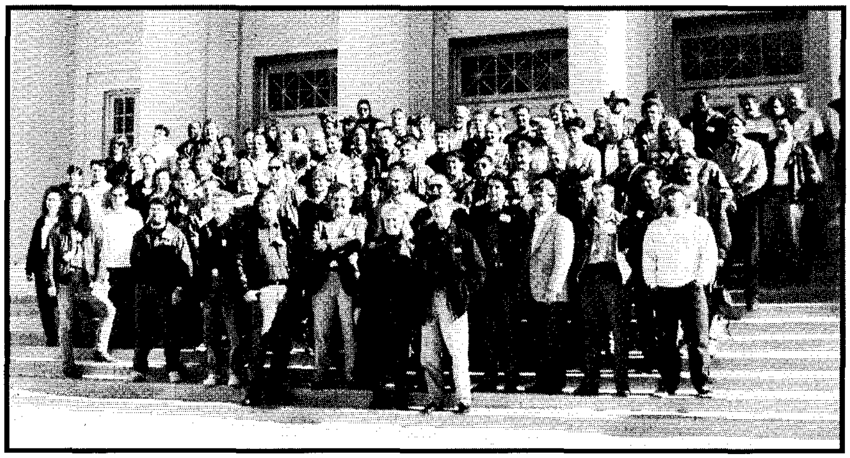
Live Sound Workshop '95

Chapman University in southern California became the site of an influx of audio people and equipment as the 1995 version of the Live Sound Workshop rolled into town. This workshop has become a favorite among touring people, as participants get to see first hand the methods used by those who have succeeded in this highly competitive end of the audio business. For some reason, this workshop always places competition in the background, and these people work together to further their craft and extend a helping hand to each other and those who attend.

The staff of the workshop included some icons from the touring business, including Will Parry, Albert Leecese, Howard Page and David Rob. Mick Whelan of JBL gave a fascinating and memorable demonstration of device arrayability, a prime concern in the touring business. Of course, the rigging course was instructed by Harry Donovan, and Craig Janssen presented a DSP primer and shared his knowledge of low-frequency arrays.

The equipment used during the workshop was staggering, both in quantity and quality. This allowed participants to get some hands on mixing experience with a live group on stage.

As with any good product, we watch the live sound workshop improve each year. This year was no exception, due largely to the support and promotion of Pro Sound News. We applaud them for their "synergy".



A Recording Made in Heaven

Those of us who think of ourselves as "Baker Street Irregulars" via Sherlock Holmes, nee A. Conan Doyle, know of the great Spanish violinist Pablo De Sarasate whose concerts, the amateur violinist, Holmes attended.

Dorian Records has outdone themselves with a "Homage to Pablo De Sarasate" recorded in the Troy Saving Bank Music Hall, DOR-90183. The artists are Rachel Barton (violin) and Samuel Sanders (piano). Miss Barton, only nineteen, already has been distinguished as Laureate to the 1992 Fritz Kreisler Violin Competition in Vienna. Mr. Sanders has been the accompanist to such notables as Itzhak Perlman, Pinchas Zukerman, Yo-Yo Ma, Mstislaf Rostropovich, Jesse Norman and a host of other distinguished performers.

The recording is definitive, the artistry memorable. Here's how piano and violin are meant to sound at the best seat in the house and what a house! Troy Savings Bank Music Hall is among the best in the world.

We've always admired the quality of Dorian recordings but this combination of composer, hall, artist and engineer has to be one planned in heaven.

Repeated playing has failed to dim my ardor for all involved. If this recording doesn't redefine how seductive a good violin and violinist can be, you need a new sound system. The 375 year old violin Miss Barton plays so exquisitely is on loan to Miss Barton from the Stradivari Society. It is our firmly held conviction that in each generation such treasures truly belong to those who can make them speak gloriously and Miss Barton and Mr. Sanders truly do so.

Do we like this recording?—Guess! *dbd*

Orlando '95

People and Places...

From left to right are Chad Bender,..., Chuck Kjergaard, Jim Guthrie, Judy Gill, and Chris Montgomery.

Rank Leisure

Rank Leisure is a company that owns and operates several high-profile night spots in the Orlando area. They are always well-represented in the Orlando class. Some of their venues include King Henry's Feast (a dinner theater), Wild Bill's (another dinner theater), and Blazing Pianos (a rock and roll piano bar). Mixing at these places is somewhat of a juggling act that includes audio, lighting, special effects, etc. It is even worse for Judy, who sometimes fills-in as a "serving wench" at King Henry's. A position at Rank Leisure requires proficiency in all of the above, as well as installation and measurement skills. This group has what it takes, as the sound quality that we heard at these sites was excellent. *pb*

Disney

Ken Corleu and Doug Wyatt represented Disney's Pleasure Island at the Orlando class. Their duties include the design, installation, testing, and mixing of the systems at the Pleasure Island clubs. Doug's credits include designs for Disney and beyond, including several Hard Rock Cafes. As one would expect, the systems at Pleasure Island are truly state-of-the-art, with top grade equipment (and lot's of it). The acoustics of the venues are given as much attention as the sound systems, which helps explain why these systems work so well. Their input during the class was invaluable, and we look forward to an ongoing relationship with these fine people. *pb*

The Absorption Coefficient of Water

Who could visit Florida without conducting some field tests to validate the absorption coefficient of water? The site that Brenda and I chose was the lake at Disney Village. We determined that the water was indeed wet, didn't have much absorption, and that further testing (next trip) will be necessary to validate the results of these tests. *pb*

JASA CD-ROM Available

The Acoustical Society of America (ASA) has answered a basic need with elegance and economy. Fundamental to any audio person's personal growth is a full technical understanding of the acoustic environment he or she is called upon to work in. A knowledge of what has and what hasn't worked in such environments in the past is essential. There are two ways one can proceed. The first is learn it the hard way by personal experience. The second far more enjoyable way is to study the giants upon whose shoulders we all stand.

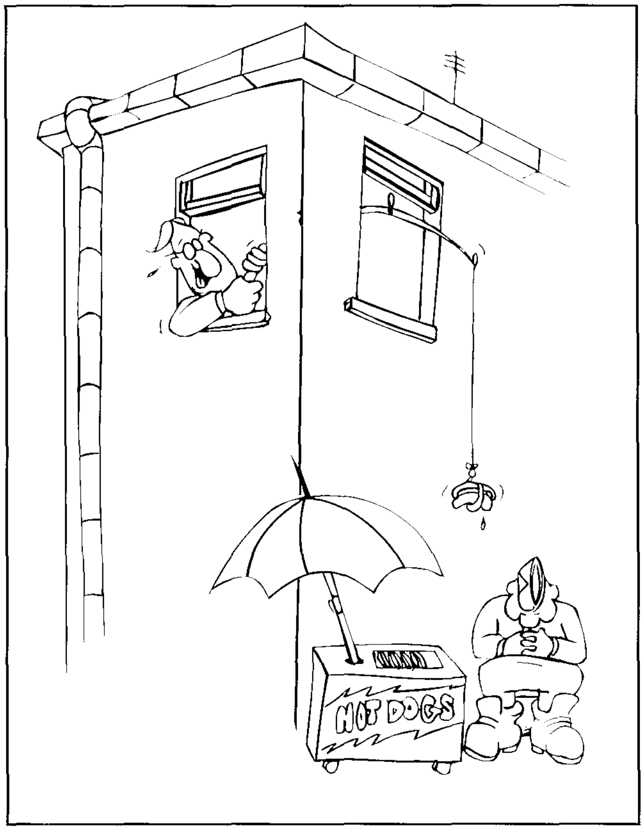
The ASA has provided a CD-ROM of all the issues of their Journal from 1927 through 1955 for $74.50, a savings of over $9,000 compared to having purchased them over the years. It is the Society's intention to continue to produce these CD-ROMs until the entire archived material is available in this convenient, searchable format. What a thrill it was to call up Harry Olson and read the articles published during the time frame of the lst CD, then print out the portion of the article you wished to use in an article being written.

Our prayer to societies and to entities such as Bell Laboratories with regard to their Bell Telephone Journal would be that they do likewise. Early Journals from AES, SMPTE, IEEE all contain invaluable resources for today's budding engineers. Not only can they be researched for great ideas that formerly were not able to be realized by that day's technological tools, but could today fuel marvelous uses of digital techniques. They also would allow the reader to see what really clear, concise, informative writing looked like compared to some of today's techno-babble.

Anyone Can Order

There is nothing on the order form that asks if you are a member of ASA, which would indicate that anyone could order the CD-ROM. I called and verified that this is true. ANYONE may order the back volumes for $74.50. I suspect that as the volumes get closer to a current date that the price will go up. I can't recommend too highly that Syn-Aud-Con grads avail themselves of these truly useful CDs and use them as the excuse for equipping your computers to handle them.

Acoustical Society of America, 500 Sunnyside Blvd, Woodbury, NY 11797-2999. Tel: 516-576-2360 or fax 516-576-2377. Check or credit card $74.50. *cpd*



Professional Services

Acoustical Consultants are provided a listing in this section. There is no charge. The only requirements are that you are a full-time consultant, that you have attended a Syn-Aud-Con seminar, and have an active subscription to the Syn-Aud-Con Newsletter. If you would like to be on our Consultants page, send in four (4) business cards for our file.

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Personnel from these manufacturers receive Syn-Aud-Con training which provides still another link in the communications circuit between the ultimate user and the designer-manufacturer of audio equipment. The are "in tune" with what a Syn-Aud-Con grad needs.

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