SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

newsletter

Volume 12, No. 1
Fall 1984

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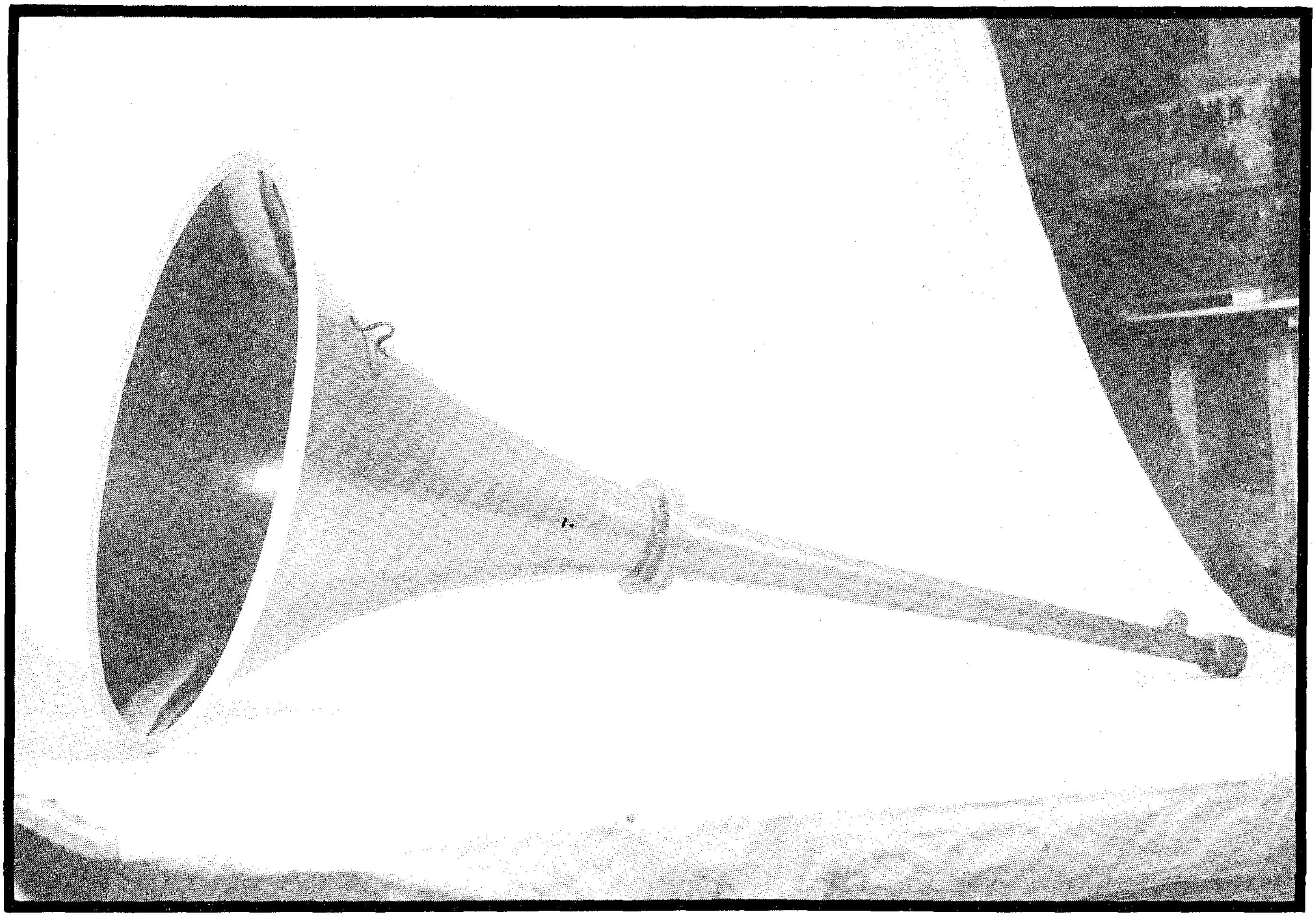
SYNERGETIC
Working together; co-operating, co-operative

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

EXCHANGE OF IDEAS

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

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



DH-2 STRAIGHT HORN - 1934 - AN ORIGINAL TOA PRODUCT

TOA ELECTRONICS, INC./1934 -

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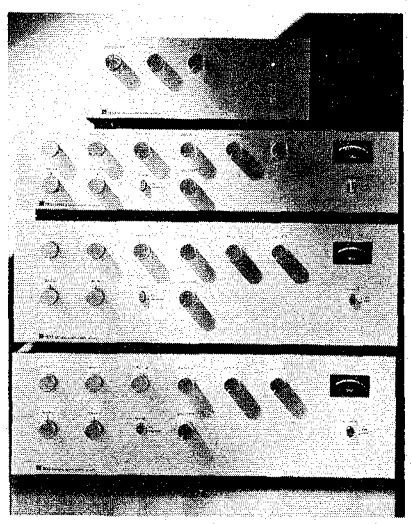
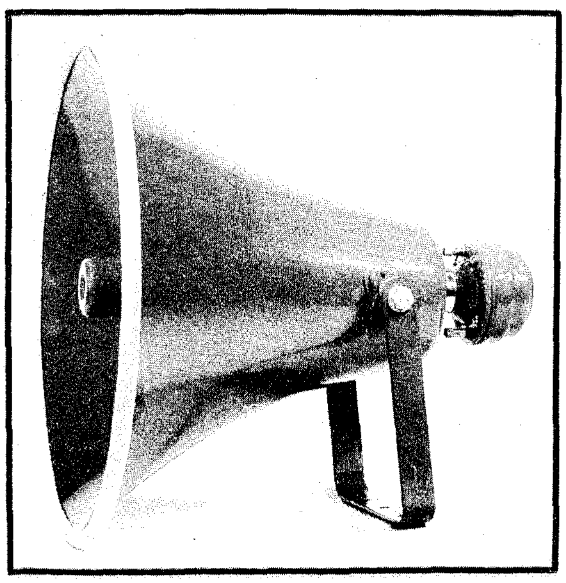
TOA ELECTRONICS, INC. 1934 -

TOA Electric Manufacturing Company came into being in 1934 to manufacture and market horn speakers, microphones and amplifiers. TOA started as a family owned and managed business to manufacture sound reinforcement equipment. TOA is still family owned and still manufactures sound equipment only -- a truly remarkable record.



TOA 900 Series Amplifier

In that fifty years, they have grown into an international audio-only company with offices throughout the world and over 4,000 different products in their inventory (about 500 of them imported into the United States). TOA is well known for their careful marketing to sound contractors in the United States and for having exceptionally well-qualified personnel truly interested in that special market place. Syn-Aud-Con is pleased to have TOA as a sponsor and looks forward to their next fifty years in our international audio industry. TOA, like Hewlett-Packard and other soundly based international companies, have truly international staffs and are part of that "melting pot" of professional businessmen that clearly see that the world is indeed a place where we all have to live together and that it can be done both harmoniously and profitably.



TOA TS-12V Re-entrant Horn - Manufactured 1952

STAFF

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Jan Kreitz
Pat Carlson

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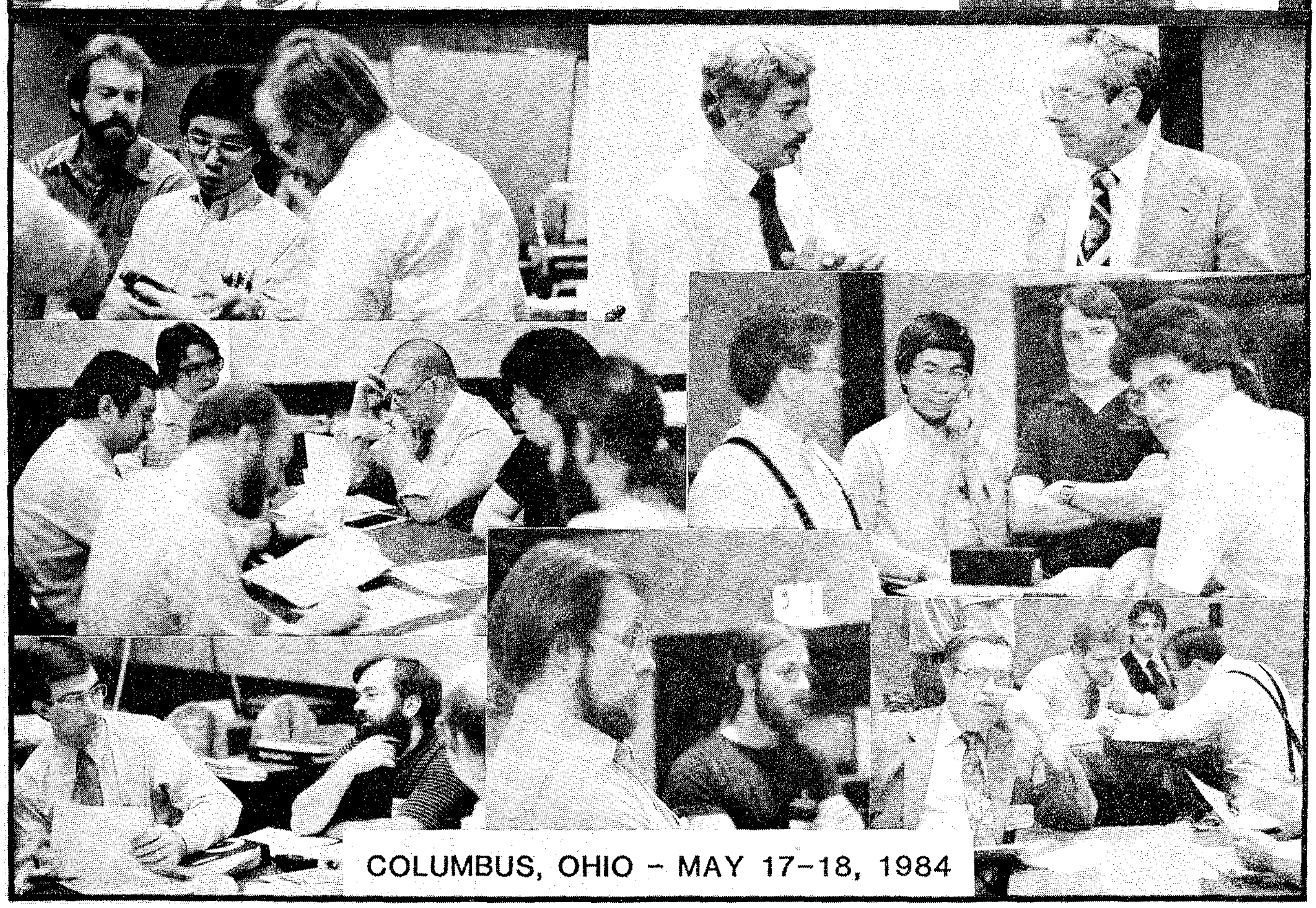
OKLAHOMA CITY - MAY 9-10, 1984



COLUMBUS, OHIO - MAY 17-18, 1984

1984/85 SYN-AUD-CON SCHEDULE

1984

Two Days With Victor Peutz - December 10-11 - Pasadena, California

V. M. A. Peutz of The Netherlands is one of the leading acoustical consultants in the world today. His work in calculating and objectively measuring the parameters that affect speech articulation has received world-wide recognition. His work in the design of concert halls is without peer at the present time. Naturally, Mr. Peutz and his organization were among the very first users of TEF® analysis. Mr. Peutz has agreed to spend two days with us sharing his latest monumental work on $\%AL_{cons}$, his latest insights into the design of concert halls (he has five in progress at the moment), and his utilization of TEF as a key tool. Mr. Peutz is employing the TEF analyzer in acoustic modeling as well as in its previously identified uses.

This meeting will be two days of unique opportunities to ask questions, and have them answered authoritively, about the acoustics of concert halls, the latest uses of TEF analyzers by such designers, and his latest progress in articulation research. We will be meeting in an outstanding American concert hall which will allow Mr. Peutz to also provide us direct instruction in how to listen in and to such spaces.

An optional third day is added at no charge for conference attendees. For more information, call (714)728-0245.

1985

Sound Engineering Seminar: January 23-24, 1985 - Hawaii*

*Note: This is a change of date from December 4-5, 1984. We think it will be a better schedule for those on the mainland wishing to attend a class in Hawaii.

Sound Engineering Seminar: February 6-7, 1985 - San Juan Capistrano, CA.

Business Organization & Management Workshop: Feb. 12-14, 1985 - San Diego, CA.

Workshop Staff: Victor Hall, Communications Company, Inc. and Harvey Earp, J. W. Davis and Company.

LEDE Control Room Design Workshop (Tentative): Early March - England

Loudspeaker Array & Sound System Design Workshop (Tentative): March - West Coast

TEF Workshop (Tentative): March - West Coast

Sound Engineering Seminar: April - Texas

Sound Engineering Seminar: April - Kansas City

Grounding for Audio Systems: May 7-9, 1985 - Minneapolis, MN.

Workshop Staff: Edward G. Lethert, Jr., of Michaud, Cooley, Hallberg, Erickson & Associates, Minneapolis, MN., and Christopher R. "Topper" Sowden, of Joiner-Pelton-Rose, Inc., Dallas, Texas.

Sound Engineering Seminar: May 22-23, 1985 - Indianapolis, IN.

Installing & Troubleshooting The Sound System: June, 1985 - Atlanta, GA.

Workshop Chairman: Philip B. Clark of Diversified Concepts, Inc., Marcellus, New York.

Sound Engineering Seminar (Tentative): June - Eastern Canada

Sound Engineering Seminar (Tentative): June - New England

We have asked the members of each workshop staff to send us material to help us prepare the workshop brochure. If you will indicate to us the workshop you are interested in, we will photocopy their material for you so that you will have a feel for the workshop content.

FEE: **2 Day Class: $400.00** **3 Day Class: $600.00**

NEW LEVEL & ANGLE INDICATOR

Don Eger showed up at the Loudspeaker Array and Sound System Design Workshop in Nashville with a new level and angle indicator that had our tongues hanging out. (Sandy Swartzendruber of Bristol, Indiana, saw it at the Consumer Electronics Show and bought several, selling one to Don Eger.)

Doug Wilkens of Peirce-Phelps was in the class and promptly arranged for distribution of this *very* useful level. Doug sent us the following information on the level:

A combination level and angle indicator offers greater accuracy and versatility than can be obtained from an ordinary level or inclinometer. This unit is especially useful for speaker alignment and rigging.

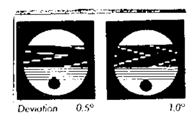
Designed to set angles and/or measure them, the Inogon Level and Angle Indicator provides direct numerical readings of angles with a precision of 0.2°. Pitch rise can be quickly converted to angles by a conversion table provided with the unit. For example, 4:1 angle is found by $\mathrm{TAN}^{-1}(1/4) = 14^{\circ}$.

The tool uses a new optical technology that eliminates parallax errors. Angle measurements are based upon a change in visual patterns that occurs when light passes through two superimposed optical windows.

Physically, the new tool consists of an angle indicator calibrated in degrees that fits into a rule calibrated in inches. The indicator frame and window are made of high-impact plastic.

General Use

Some uses will not require a high degree of precision, and small deviations may be acceptable. In those cases, it is not necessary to obtain a pattern of parallel lines in the window, as each pointed arrow represents a deviation of only 0.5°.



To set angles, an adjustment screw on the indicator is turned to align the desired angle (in number of degrees) on a moving scale with a zero reference point on a fixed scale.

Then the tool is tilted until parallel lines are seen in the window of the angle indicator, showing that the tool is now inclined at the desired angle. If a number of speakers are to be set at the same angle, the tool can be used over and over again without moving the adjustment screw. This eliminates the time-consuming process of re-aligning an ordinary level and protractor for each speaker or horn.

To measure an angle, the Inogon Level and Angle Indicator is placed on or against the surface of the speaker cabinet or mounting frame. The adjustment screw is rotated until parallel lines appear in the indicator window. The angle of the pitch or yaw is then read out on a scale in degrees. Additionally, the tool allows simple computation of heights (using basic trigonometry formulae) of objects which cannot easily be measured.

The Inogon Level and Angle Indicator is available in lengths of 10, 24, and 48 inches, magnetic or non-magnetic. Prices are listed below:

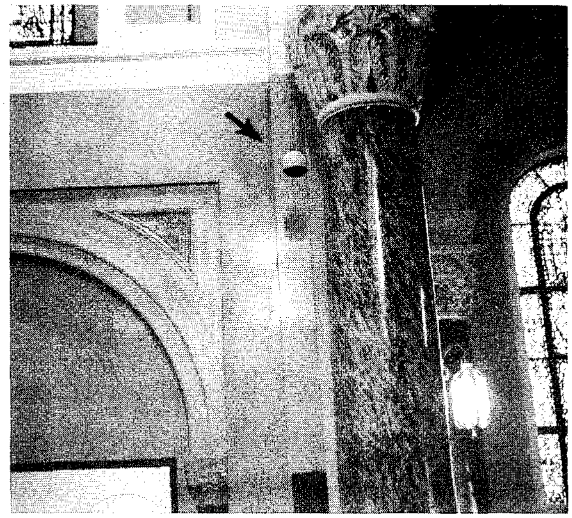
MODEL NO.	BASE RULE SIZE	BASE RULE COMPOSITION	PRICE
3252-BS	10"	Black Anodized Brushed Aluminum with Numeric Scale	$51.45
3253-MBS	10"	Black Anodized Magnetic Brushed Aluminum with Numeric Scale	$61.45
3602-BS	24"	Black Anodized Brushed Aluminum with Numeric Scale	$55.45
3603-MBS	24"	Black Anodized Magnetic Brushed Aluminum with Numeric Scale	$67.45
3122-BS	48"	Black Anodized Brushed Aluminum with Numeric Scale	$61.45
3123-MBS	48"	Black Anodized Magnetic Brushed Aluminum with Numeric Scale	$73.95

The Inogon Levels are available from: Peirce-Phelps, Inc., "Audio Systems Division"
2000 North 59th Street, Philadelphia, PA 19131

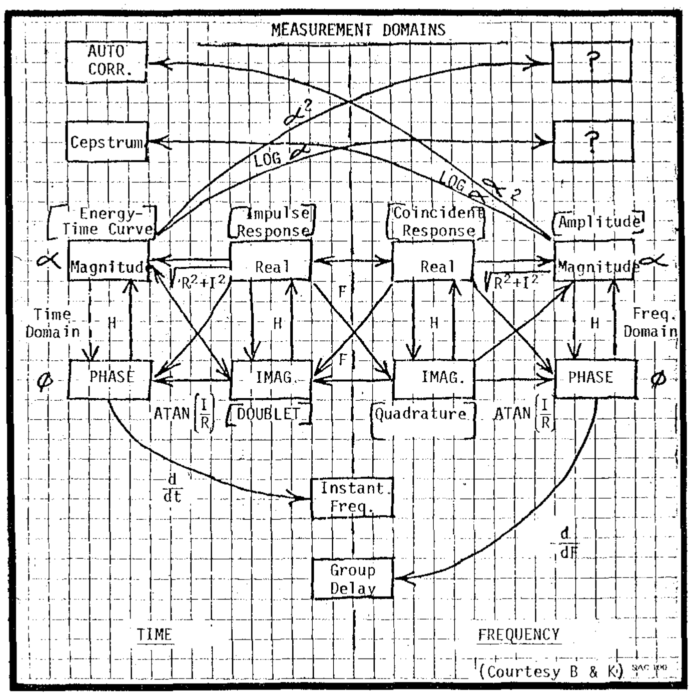
♦

CREATIVE DISTRIBUTED SYSTEM

Our Lady of the Angels Church in Brooklyn, New York, is a beautiful edifice. Charles Bilello overcame the classic conflict between Q_{avail} and D_{2max} by using J. W. Davis & Company Spheres unobtrusively overhead with Crown amplifiers, IRPI automatic mixer, Shure microphones, UREI ANCA, and processing equipment from DBX. Charles, who does not accept mediocre performance from anything, says it's an excellent installation that totally satisfied his customer. ♦



WHERE DO WE GO FROM HERE?



TEF® analyzers are now out in the field in almost one hundred different user's hands. Most of us are fully engaged in just trying to keep up. Heyser has said there are many more measurement domains than just time and frequency. We now have tools available to search for them. The positron-emission tomogram showing the distribution of F^{18} labeled 2-deoxyglucose in the brain coupled to TEF analysis of the cause and effects observed with different auditory stimuli is one possible path to the discovery of new domains and different manipulation of the present domains. Just as impulse response had hidden in it the magnificent detail of an ETC plot so too it's reasonable to suspect that hidden in our present measurements is the kernel of still newer and better ways.

What might lie along the other axis of the domain chart? Where does the musician's note on the score "with feeling" fit into our measurements? Do we "hear" potential energy? One telephone engineer recently described a signal as several bursts of ringing with "bursts of silence in between". What is the measurable state of infinite S/N? What is the measurement of relevant experience? We've all had the experience of re-reading a book about some given place both before and after visiting it and realizing it's two different books as a result of the real exposure to the environment. The entire Sherlock Holmes series is based on Holmes in one domain and Watson in another with only Holmes able to "transform" the data from one to the other at the end of each story.

The future of the TEF and of audio lies in the Holmes domain. Don't be a Watson. ♦

THE ANALYTIC SIGNAL

The TEF® analyzer employs the analytic signal developed by Dennis Gabor back in 1947. It is the most powerful mathematical model we have for an audio and acoustic system signal source.

The analytic signal shown at the top of Fig. 1 is a steady state signal with time t as its center axis and the rotating phasor tracing out the helical pattern shown. At a 90° angle from each other (a Hilbert transform) we find the real part of the phaser and the imaginary part of the phasor. If we plot each point traced out by the phasor for the identical time, we'll find that when the real part is at a maximum, the imaginary part is at a zero and vice versa. This is because energy lying on the real axis is pure potential energy and energy lying on the imaginary axis is pure kinetic energy. The magnitude of these two vectors (the magnitude of the signal) is found by:

$$\text{Mag} = \sqrt{R^2 + I^2}$$

Note that in the illustration the magnitude is depicted as the same over a *time* axis.

The phase of the signal is found by:

$$\text{phase} = \text{Tan}^{-1}\left(\frac{I}{R}\right)$$

for any given point in time.

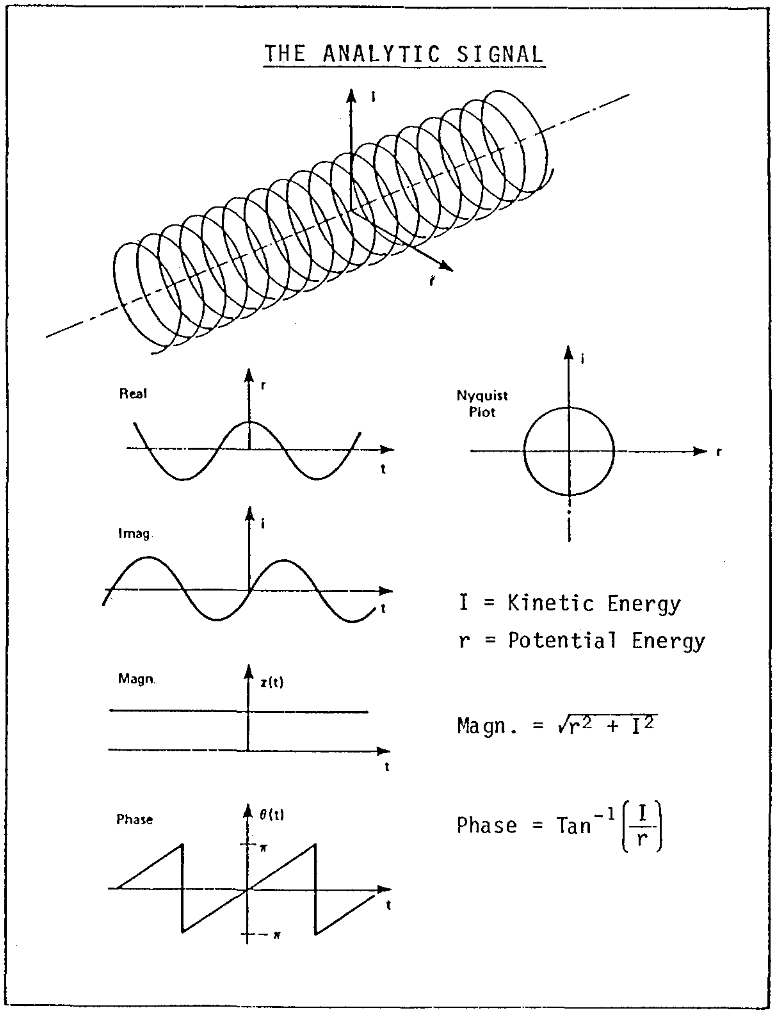


Illustration of the analytical signal for a cosine wave.

The Nyquist Plot

The end view of the analytic signal is called the Nyquist Plot after the famous Bell Telephone Laboratory scientist H. Nyquist. It is necessary to remember that you are viewing all of the rotations even though they are masked by the latest rotation. The vertical axis is the imaginary axis and in acoustic measurements represents the kinetic energy. The horizontal axis is the real axis and in acoustic measurements represents the potential energy. On a TEF analyzer when the cursor is brought onto the screen the notations on the left side of the screen include the real part, the imaginary part, the magnitude, and the relative phase angle (relative to the zero time of the chosen signal delay) as well as the frequency at which each of these values occur. Both the real and imaginary parts appear at their own set of terminals at the rear of the analyzer.

Impedance Measurements

The TEF analyzer has no way to detect what the input signal actually is, such as, is it electrical or is it acoustical? Therefore, connecting it across a transducer in a constant current impedance measuring setup allows the measurement of the complex impedance. The vertical scale is reactance, the horizontal scale is resistance, the phasor length is the impedance, and the phasor angle is the phase angle of the impedance.

A Fundamental Observation

If the system is linear, the real and imaginary parts are Hilbert transforms of each other. In non-linear or dispersive situations, an actual measurement of the phase as well as the magnitude becomes necessary because they are then no longer related by the Hilbert transform. We need to carefully distinguish between the transient response of a system measured as a steady state response to a transient input versus the transient response of the system as its "natural mode" rather than its "forced mode." Remembering that *"phase" is a steady state phenomenon* helps to understand the search for true signal delay mechanisms in the system. ♦

SMILE

Launegayer's Observation: *Asking dumb questions is easier than correcting dumb mistakes.* ♦

SIGNALS IN QUADRATURE – Real and Imaginary Components

The Tecron TEF® analyzer processes audio signals in quadrature. That is, the signal is looked at from the real part and the imaginary part of the total signal energy. The "real part" represents the *potential energy* and the "imaginary part" represents the *kinetic energy*.

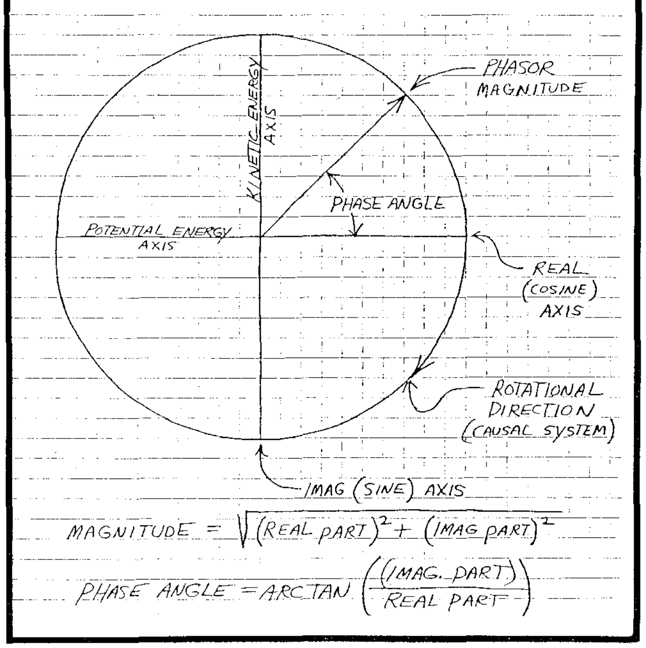
The Nyquist (NPP) display allows us to view this partitioning of energies. The energy time curve (ETC) display allows us to view their combined effect. Phasers (the point tracing out the Nyquist plot) also allow us to view magnitude and phase.

If we have a pair of conjugate phasers, i.e., rotating in opposite directions, then the real and imaginary parts can be thought of as the resultant of the two. Two phasers rotating in opposite directions suggest the concept of a negative frequency - negative to the initial point of reference. The use of negative frequencies is a convenient way to describe a real-valued function using pairs of complex-valued functions.

The TEF® analyzer has both the real (cosine) and the imaginary (sine) signals available at separate outputs on the rear of the analyzer.

The NPP of a straight wire is a point displaced on the real axis by the magnitude of the signal and coming straight out of the screen toward the viewer. The phase delay of the system causes this point to rotate around the circumference of a circle. The more delay, the more circle that appears. Devices with no significant delay produce a circle that rotates around the original displaced point while a few hundred microseconds of delay produces a full circle rotating around the center of the screen.

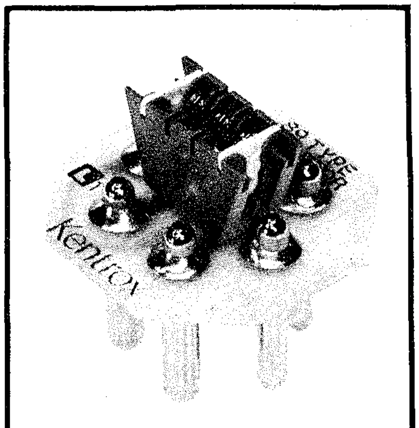
The Nyquist Phase Plot



The NPP is also dependent upon the chosen T_R of the analyzer. If you examine the NPP of the Sunn signal delay unit with a 10 msec window, 1.0 msec mistuning causes a full circle on the NPP. If the T_R is 1.0 msec, then a 0.1 msec mistuning causes a full circle. Great care must be exercised to differentiate between signal delay and phase delay in making measurements and we hope that Dick Heyser, Gene Patronis and Gerald Stanley can produce a series of example measurements sampling from expected extremes through likely cases clear out to illusions that can occur. ♦

K-PAD TO 1C ADAPTER

The following is an excerpt from Plantronics/Kentrox's publication, "Bright Spot #11":



K-Pad to 1C Adapter

"Almost every central office has one or more bays of 1C-type (or equivalent) attenuator panels. Having the right values of 89-type resistors on hand and ready for quick attenuation changes can be a problem, particularly for massive conversions such as switch changes. Kentrox has the solution. Our line of K-Pads covers the attenuation range from 0.0 dB to 35.0 dB (plus infinity) in 0.2 dB increments. These K-Pad modules can be installed in a 1C-type panel using our Model 45007 plug-in adapters. Once the plug-in adapters are installed, you can use any of more than 120 values of K-Pad attenuator modules to get just the amount of attenuation you need. These modules are electrically equivalent to 89-type resistors. Since central offices already extensively use K-Pads in other mountings, eliminating 89-type resistors also means stocking only one type of attenuator. A supply of K-Pads uses about 1/10th the storage space required for 89-type resistors. And, K-Pads are normally shipped "off the shelf."

Kentrox Capabilities Brochure

"Kentrox is a company of people dedicated to serving people -- our customers -- that's the most significant 'Kentrox Difference.' For a look at Kentrox from the inside, get a copy of our new Capabilities Brochure. It explains 'the Kentrox Difference' and describes the capabilities that make Kentrox a leading supplier in the telecommunications industry. The 8-page brochure is attractively printed in four colors."

The brochures mentioned above are available from Plantronics/Kentrox, 14375 N.W. Science Park Drive, Portland, Oregon 97229. Their telephone number is (503) 643-1681. ♦

A TEF® UPDATE

Dick Heyser has been busy making his own versions of the TEF disk. We recently received version MK 1.1 from him (his second to date).

Heyser's New Disk

Dick's new program *adds* to the conventional TEF disk the following functions without losing any of the previous functions:

1. Vector and scalor averaging.
2. Screen storage and screen differencing.
3. Linear or log amplitude.
4. Group delay.
5. 12 bit digital oscilloscope.
6. 12 bit digital FFT.
7. Harmonic distortion analysis to the 9th harmonic with any signal delay to 240 secs, including the phase of the harmonic.
8. New expanded ETC (full screen 2.5 msec.).
9. A test oscillator for steady tones at any desired frequency and amplitude.
10. A sawtoothed oscillator.
11. Impulse response.
12. Doublet response.
13. Keyboard channel selection.
14. Decimal - binary expansion.
15. Quadrature toggle (i.e., Hilbert transform).
16. A technique for using the 3-D display to store 32 different measurements if desired.
17. Cursor commands that go directly to frequency or time chosen.
18. A quick entry mode.
19. Ability to change gains from the keyboard during tests.
20. Excellent documentation or the disk written by Heyser.

Dick informs us this is just the beginning of his programming efforts. This new disk allows precision "one port" measurements to be made in the classic manner and at the same time having the enormous advantage of TEF analysis for all two port measurements.

TEF vs FFT

It is a simple matter of *fact* that anyone using dual channel FFTs for two port measurements of acoustic transducers doesn't understand the fundamentals of measurement. FFTs cannot be used for:

1. Non-linear systems.
2. Dispersive systems.
3. System analysis requiring both a short time window and a good frequency resolution.

The greatest manifestation of ignorance currently prevalent in audio today is the attempts to duplicate TEF measurements with *any kind of FFT*.

The Heyser Transform

The Heyser transform is not yet published; therefore, those few individuals who truly do understand the use of the classic Fourier transform and how it is actually employed in the discrete Fourier transform must have faith in Dick. Dick states that the Fourier transform is only a limited case of his more general transform. Those of us actually using the best of both types of machinery already know which technique is the general case by the results obtained.

The large number of "they *think* they understand the Fourier transform" will continue to mislead, misinform, and mismeasure until the Heyser peer group swells sufficiently to allow academia to "discover" this new transform. It must be hard to be a Heyser surrounded by dull ears, blind eyes, and closed minds but his peer group is enlarging and we believe he'll receive the credit due him one day for his truly monumental work in the development of a vastly superior mathematical model of acoustic reality. The mini-minds of the world give Silver Medals to work deserving of a Nobel prize, but then, that's the very definition of a mini-mind.

New TEF-10 Owners

Mr. Bob Richards
McClear Place Studios
225 Mutual Street
Toronto, Canada
M5B 2B4

Mr. Neil Grant
Discrete Research-Unit 15
No. Field Industrial Est.
Beresford Ave., Wembley,
Middlesex HA0 1YB ENGLAND

Mr. Horace Wee
Audiotek(S) PTE.LTD.
8 Jalan Antoi
Seletar Hills Estate
Singapore 2880

Mr. Kaj Gamback
SEA-LAB
Handelsvagen 44
12238 Enskede
SWEDEN

Mr. Dan Zellman
Schwartz Recording
420 Lexington, Ste. 19
New York, NY 10017

Mr. Ted Vandenheuvel
Maryland Procurement
9800 Savage Road
Attn: R912
Fort Meade, MD 20755

Mr. Gene Kurzweg
Score, Inc.
1333 South Birchlake Blvd.
St. Paul, MN 55110

Mr. Gary Kendall
Northwestern Univ.
School of Music
711 Elgin Road
Evanston, IL 60201

Altec
10500 West Reno
Oklahoma City,
OK 73128

OMNI Hearing Systems
P. O. Box 115008
Carrolton, TX 75006

♦

SOME VIEWS OF EQUALIZER SIGNAL DELAYS

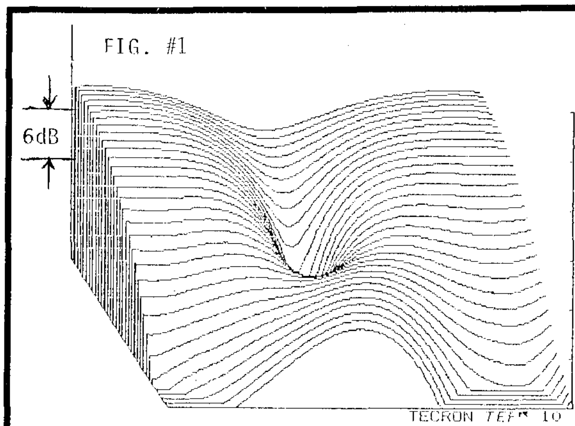
We live, of course, in a causal universe so far as we can presently detect with our limited senses and finite scientific extensions of them. Therefore, we expect devices to have delays inherent in their operation. "Instantaneous" is a metaphysical concept, not a physical term.

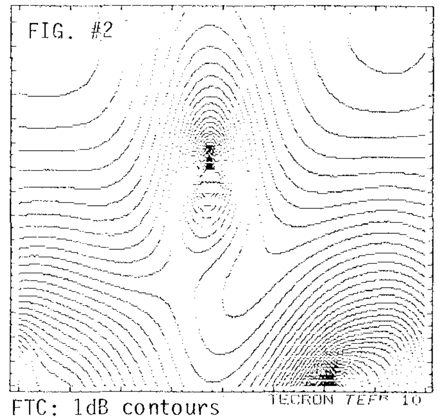
A Classic Example

Figure No. 1 is the 3-D TEF® measurement of a RCL filter designed by Art Davis and Don Davis back in 1965. Note the scales. They are the same for all 3-D measurements discussed here. Figure No. 2 is the frequency time curve FTC of the same unit. It is clearly evident that:

1. It took a finite length of time for the notch to appear.

2. Early energy passing through this circuit is wide band (short in time) and a much higher level than the steady state measurement of the same filter. Just as interesting is the fact that the transient depth of the notch is much greater than the steady state response as well (usually twice as deep as the steady state condition).





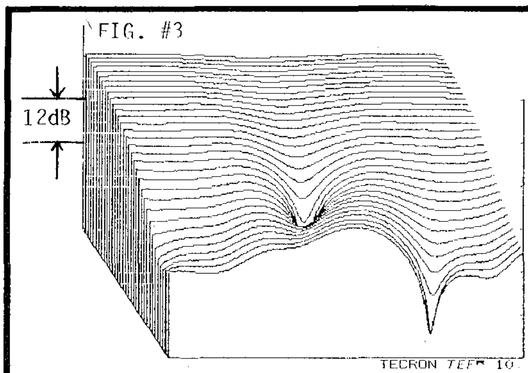
FTC: 1dB contours

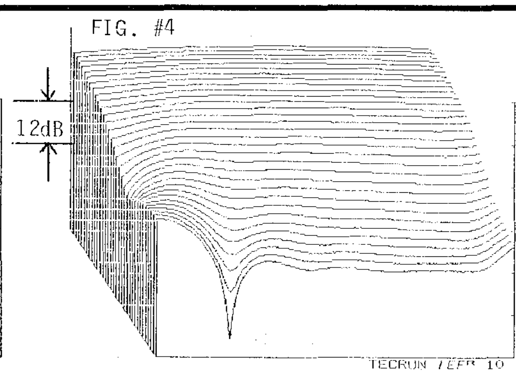
Horizontal: Auto 0.00Hz to 2000.24Hz
scale: 546.89Hz/inch or 215.31Hz/cm.

Resolution: 1.0527E+00 Feet & 1.0735E+03Hz

Time of test: 1500 microseconds 1.6950E+00 Feet (front)
to 0 microseconds 0.0000E+00 Feet (back)
-48 microseconds/step or-5.467741935483E-2 Feet

Sweep Rate & Bandwidth: 10734.80Hz/Sec & 1.0000E+01Hz





Horizontal: Auto 0.00Hz to 2000.24Hz
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to 0 microseconds 0.0000E+00 Feet (back)
-48 microseconds/step or-5.467741935483E-2 Feet

Sweep Rate & Bandwidth: 10734.80Hz/Sec & 1.0000E+01Hz

A Very Narrow Band Passive Filter

Note first that all time parameters are identical to the first case but the dynamic range had to be increased in order to see the amplitude effects at a longer signal delay time. This filter took a great deal longer to operate and the higher Q allows a very wide range high amplitude signal to pass through ahead of the filter's effect. Note the unexpected additional notch (Figure No. 3). When looked at on an even longer time scale, this filter exhibited a series of such extraneous filters on either side of the "ringing."

An Active Filter

A final example is that of an active filter. This one barely made it into our arbitrary time scale. (See Figure No. 4)

Conclusion

The transient response of equalizers can vary dramatically when viewed in 3-D TEF displays. Many previously unexplainable differences between what was measured by the older conventional measurements and what listeners clearly heard become identifiable when the full 3-D response is viewed. ♦

EXPLORING THE ACOUSTICS OF A GREAT THEATRE

From Peter George Associates, Inc., New York, NY:

During its nearly sixty year life, the Ohio Theatre (Columbus, Ohio) has been filled with the sounds of Cab Calloway, Kate Smith, Jean Harlow, Fred Waring, Ginger Rogers, Judy Garland, Laurel and Hardy, Debbie Reynolds and, more recently, the Columbus Symphony Orchestra, Bruce Springstein, Beverly Sills, Tony Bennett, Isaac Stern, Carol Channing and the Who. However, the grand old lady was recently insulted with electronic sound that even John Cage would find dissonant.

As part of a program to design and build a new orchestra shell for the Ohio Theatre, it was decided by consultants Peter George Associates to use the most up-to-date technology available to establish some base line parameters, with the existing inadequate orchestra shell in place, against which the benefits of the new shell could be measured.

Using the recently available Crown Tecron computer acoustical analysis system (operated by Charles Bilello, an original TDS licensee) millions of bits of information on the Ohio Theatre acoustics were recorded on the floppy discs that the computer uses.

At the completion of this study, Peter George reported that 'For the first time we have been able to establish the mostly good-to-excellent acoustics of this theatre with technical back-up. Until now, the acoustical reputation of the house has been based only on opinions, some of which have been contradictory'.

The orchestra shell, being built by the Wenger Corporation, will be in place, tested and tuned, in late September, ready for the new season of the Columbus Symphony Orchestra, led by maestro Christian Badea. ♦

SMILE

McDonald's Corollary to Murphy's Law: *In any given set of circumstances, the proper course of action is determined by subsequent events.* ♦

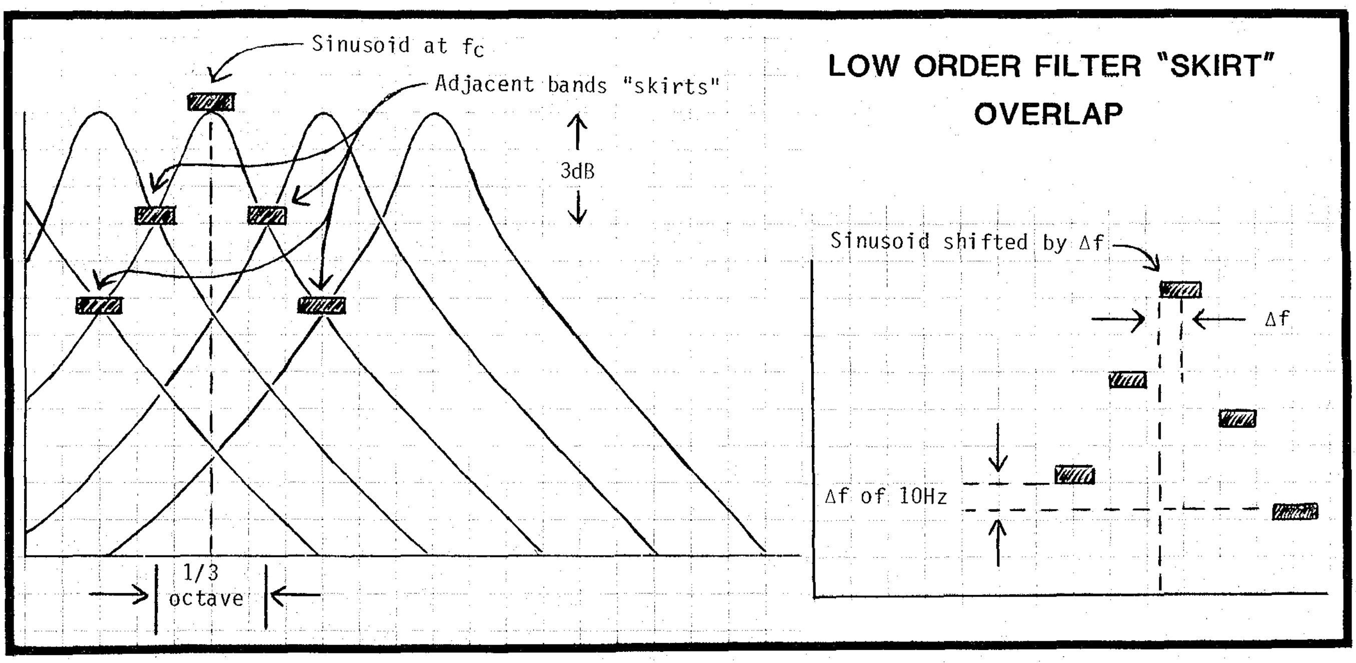
REAL TIME ANALYZERS FOR AUDIO

1/3 octave (actually 1/10th decade) analyzers (frequently mislabeled as *third* octave analyzers) are highly useful everyday work tools for the adjustment of audio systems. These devices are constant *percentage* bandwidth (CPB) analyzers that utilize bandpass filters with bandwidths that are 23% of their center frequency (f_c)

The filters employed can have "skirts" that attenuate unwanted signals by 6, 12, 18, 24, 36dB or even greater rates per octave (i.e., dB/oct). These analyzers are used with random noise generators of the constant power per octave type (i.e., "pink" noise) that on an energy per Hz basis show *decreasing energy* with increasing frequency. A constant energy per octave signal observed on a constant *percentage* bandwidth analyzer is displayed as a uniform frequency response.

A Common Error in the Selection of a 1/3 Octave RTA

If the only input to a 1/3 octave RTA were random noise signals, then the highest resolution would be obtained by using the highest order filters (36dB or greater per octave). However, oscillators and sound system acoustic feedback signals are often the input signal in actual fact. When such is the case, 6dB/oct filters allow the widest range of frequencies to be observed to the greatest resolution above the ambient noise level (usually with 10 Hz in the 1000 Hz region). It is for this reason that we recommend the analyzers with the simplest filters for *sound system* measurements. (See the figure shown below.)



The higher order filters have their place in noise control measurements but are both less effective and far more costly when applied to sound system work.

Next to those using the decibel scale on voltmeters to read what they think is the level, 1/3 octave analyzers are the next most frequently misused measurement tool. ♦

SLIDE RULE SHORT CUT

James M. Hubbard, Vice President of Atomic Sound, Inc., Oklahoma City, showed us a real "short cut" in calculating loudspeaker efficiencies.

1. On Scale 23, place the 4 ft. 1 watt sensitivity value $L_{p(sensi)}$ opposite the Q for the frequency of interest (usually 2 KHz) on Scale 24.

2. On Scale 22, the right hand arrow above the window points at the L_W (labeled dB-PWL on the rule) for a Q = 1.

3. On Scale 23, place 120 dB (the 100% L_W for one acoustic watt) opposite 100 on Scale 24.

4. Locate the L_W from Step #2 on Scale 23 and below it on Scale 24 is the percent efficiency.

We were impressed that Jim worked out this short cut without reading our instruction sheet for the slide rule, which showed he was thinking through the problem with real understanding. ♦

QUADRANT ROTATION

Quadrant rotation in audio is a Hilbert transform. Here's a Quadrant rotation in optics.

To properly view this illustration, look at it with the left side nearest you and the right side farthest from you (at right angles to the normal viewing plane). Just before you can only see the edge of the paper, the perspective should be restored. Go slowly as you rotate and you'll see it all fall into place. "The ancients keep stealing our inventions." ♦

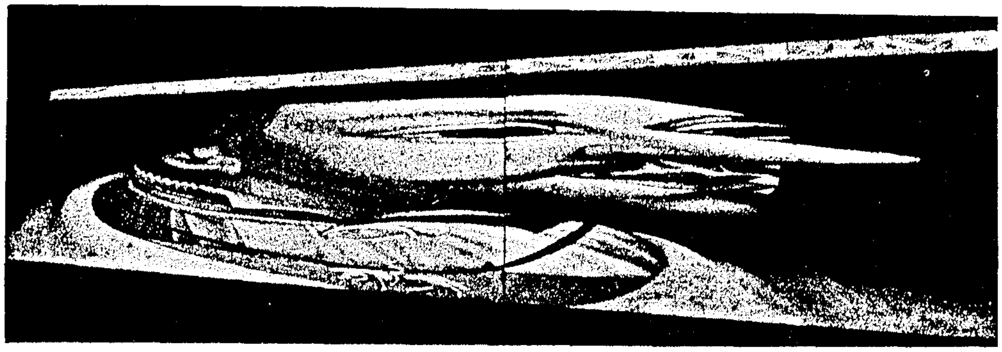


Figure 1. German School, 16th Century: Charles V, 1533.

Collection, Jacques Lipchitz, Paris

POWER EXPRESSED IN WATTS

It is stated in a book on architectural acoustics that:

> *Power is a basic quantity of acoustical and electrical energy.*
>
> *Although they are both measured in watts, they are different forms of energy and cause different responses.*

Unfortunately for that writer the statement is not true.

> *The rate at which work is done is called* *power*.

How far a mass is moved in a given time span can be expressed in the metric (SI) base units in terms of KG (mass), M (distance) and S (time).

$$W = FD$$

Where: W is work
F is force
D is distance

When we do *work* we are said to expend *energy*.

Force in metric (SI) base units is:

$$F = \frac{KG \cdot M}{S^2}$$

Distance in metric (SI) base units is:

$$\frac{M}{1}$$

Therefore: $\text{Work} = \frac{KG \cdot M^2}{S^2}$

THE ELECTRICAL AND ACOUSTICAL WATT IN METRIC (S.I.) BASE UNITS

THE ELECTRICAL WATT IS DEFINED AS:

$$W_e = \frac{E^2}{R} \quad \text{(OHMS LAW)}$$

$$E = \left(\frac{M^2\,KG}{S^3\,A}\right) \qquad R = \left(\frac{M^2\,KG}{S^3\,A^2}\right)$$

$$W_e = \frac{\left(\frac{M^2 KG}{S^3 A}\right)^2}{\left(\frac{M^2 KG}{S^3 A^2}\right)} = \left(\frac{M^4\,KG^2\,S^3\,A^2}{(S^3)^2\,A^2\,M^2\,KG}\right) = \frac{M^2\,KG}{SEC^3}$$

THE ACOUSTICAL WATT IS DEFINED AS:

$$W_a = \frac{(2\pi f\,A_{RMS}\,\rho c)^2\,(\text{SURFACE AREA})}{\rho c}$$

$$2\pi f = \left(\frac{1}{SEC}\right), \quad A_{RMS} = \left(\frac{M}{1}\right), \quad \text{SURF AREA} = \left(\frac{M^2}{1}\right)$$

$$\rho = \left(\frac{KG}{M^3}\right), \quad c = \left(\frac{M}{SEC}\right)$$

$$W_a = \left[\frac{\left(\frac{1}{SEC^2}\right)\cdot\left(\frac{M^2}{1}\right)\cdot\left(\frac{KG}{M^3}\right)^2\cdot\left(\frac{M}{SEC}\right)^2\cdot\left(\frac{M^2}{1}\right)}{\left(\frac{KG}{M^3}\right)\cdot\left(\frac{M}{SEC}\right)}\right] = \left[\frac{KG \cdot M^5}{SEC^3\,M^3}\right]$$

$$= \frac{M^2\,KG}{SEC^3}$$

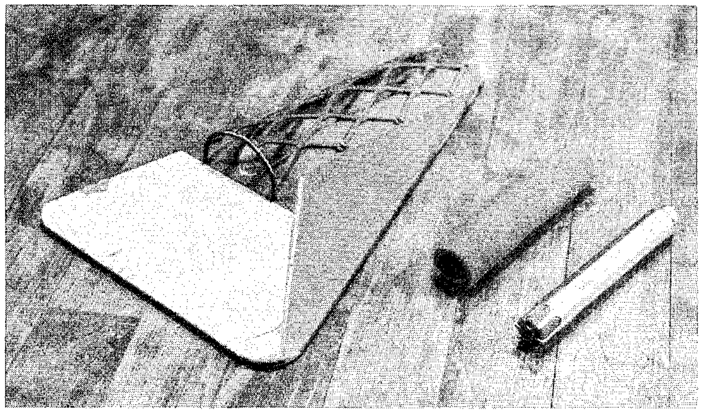
Power is the *rate* at which work is done so:

$$\text{Power} = \frac{\text{Work}}{\text{Sec.}} = \frac{KG \cdot M^2}{S^2 \cdot S} = \frac{KG \cdot M^2}{S^3}$$

The calculations shown here compute both the electrical watt and the acoustical watt into metric (SI) base units. It is apparent that electrical, acoustical, mechanical, chemical, etc., power (watts) all do exactly the same thing. ♦

HISTORICAL FORERUNNER OF THE PZM®

We first encountered "flush mounting" of microphones back in 1968 at an audio workshop at BYU in Provo, Utah. Bill Blanton told us about their use in the then relatively new L.A. Music Center. They had been specified by the consultant Paul S. Veneklasen to be placed in boxes in the stage flooring and brought up flush with the stage.



During the Nashville Loudspeaker Array Design Workshop in June, Jake Ewalt of Iowa State Center, Ames, IA, remarked that he had some of the same type units at his auditorium. We found the following letter and the photograph fascinating.

Jake Ewalt (L) with Tom O'Brien of Rauland-Borg at the Loudspeaker Array Workshop - Nashville.

At the Nashville workshop you asked for photos of Paul Veneklasen's flush-mounted floor mics. I'm afraid I threw out the mic holders last summer when we replaced the stage floor, but I found a copy of the drawings that may be of some interest... The date on the print is March 4, 1966. (Editor's note: Unfortunately we were unable to reproduce the drawings.

The mic that was specified and used in our installation was a Sennheiser MKH 405, a cardioid condenser. The actual holders that we received consisted of a 1" thick wooden plate that was set flush into the stage floor instead of the metal plate shown in the drawing. An addition that I found useful was to stuff the concrete cavity under the mounting plate with absorption to help kill the resonances and leakage from under the stage.

I am including pictures of the mic mounts that were provided as an alternative to the floor pockets, for use when the pocket positions were not appropriate (almost always). They are made of mahogany with an expanded metal protector over the microphone. The mic, an MKH 405 again, is inserted into the foam-wrapped fiber tube and the tube is held loosely under the metal screen. The whole assembly sits on 1/8" foam. Although they compromise the flush-mounting idea, they are great for protecting the mic if you have to place one in a "dangerous" area. You can see from the picture that this one has taken some hits. I don't know what happens to the cardioid pattern in this mounting (or, for that matter, in the original flush design) but they do a reasonable job of discriminating against sounds from the rear, like an orchestra pit.

I believe it was Pete Alward from Portland who also mentioned using Mr. Veneklasen's floor mounts. Maybe he can get photos of the real thing.

Thank you again for a very stimulating workshop. It's good every so often to take the old brain cells out for a stretch! Jake Ewalt

From 1968 on we regularly demonstrated the floor mounting of microphones using the 1/3 octave analyzer we had in our possession. Various manufacturers have reinvented this idea over and over again in the ensuing years. It was Ed Long that analyzed the flaws in this approach and this led directly to the PZM concept. We have been interested to note that several new versions of microphones with their diaphragms mounted at right angles to the boundary have again made their appearance albeit with smaller dimensions (good), but with directional capsules (questionable), and with unspecified pressure characteristics (bad). Again, as with all directional microphones, the response will change when placed beyond D_C.

Our special thanks to Jake for sharing this bit of history with us. ♦

EXCELLENT SPECIFICATION SHEETS

We recently received from Rauland-Borg Corporation a series of new specification sheets on their latest constant directivity horns $40^\circ \times 20^\circ$, $60^\circ \times 40^\circ$, and $85^\circ \times 40^\circ$, plus a series of constant beamwidth devices $90^\circ \times 58^\circ$, $60^\circ \times 40^\circ$, and $90^\circ \times 60^\circ$.

The constant directivity series specifications are an engineer's dream: full polar plots; DI and Q plots; isobars every 3dB to -15dB; and full mechanical mounting information. We are always pleased to see quality products fully specified from a manufacturer dedicated to the sound contractor business. ♦

MINNEAPOLIS - MAY 22-23, 1984

DENVER - JUNE 5-6, 1984

CORRECTION & ADDITIONAL INFORMATION ON ANCA

a URC company

United Recording Electronics Industries

August 23, 1984

Don Davis
Syn-Aud-Con
P.O. Box 669
San Juan Capistrano, CA 92693

Dear Don:

I have just received my copy of the Syn-Aud-Con Newsletter Vol. 11, Number 4 and, as usual, it is packed with more information than I can absorb. Congratulations on a continuing excellent job of disseminating information which would not otherwise be available to as wide an audience.

I would like to point out a possible point of confusion with regard to your discussion of the UREI Model 950 Ambient Noise Controlled Amplifier. You describe the product as having a circuit that "'locks up' the level that was present just before the system was seized for the announcement." That is incorrect. The contact closure switches the unit to a fixed gain referred to as "preset gain" which does not change with ambient noise. This feature would likely be used for a specific paging microphone located very near a loudspeaker. An installation like that could be subject to system feedback if a high noise level caused an automatic increase in gain. This circuit therefore disables the automatic level adjustment and transfers gain control to a separate front panel pot which is adjusted by the system installer for an appropriate level without feedback.

Your caution regarding the power handling capability of the system is well taken. It's entirely possible to put together a system which is nice and efficient and under no noise conditions might require, say, 10 watts of actual amplifier power. It is also entirely possible to forget that if the ANCA raises the gain by 20 dB (its maximum) that the system will then require 1000 watts of amplifier power (and speaker power-handling capability). And that's presuming average power, not peak.

Perhaps some of your readers may not immediately perceive the need for raising the system gain by 20 dB, an admittedly large amount. For those skeptics I suggest an excursion with sound level meter in hand to a baggage claim area at an airport or perhaps the local pub before and during the happy hour. All in the interest of science, of course.

Of course, there may be other more practical (read: available funds) reasons why 20 dB of increase is too much. We do remember that for every 3 dB increase in power we have doubled the power capacity needed from our amplifiers, wire, 70 volt transformers if used, and loudspeakers. It is possible that in some installations a 20 dB increase in gain, and the increase in system size and cost, even if actually called for by the environmental conditions, will not be acceptable. We do have a service bulletin available describing a simple modification which can be made to the ANCA to restrict the maximum expansion to 5, 10 or 15 dB instead of 20. The service bulletin is available on request.

Thank you for plowing through all of my thoughts. I hope I have clarified rather than muddled the situation. Please let me know if you have any questions on this or any other UREI product or the application thereof.

Yours truly,

UREI

John D. Groper
Project Engineer

Universal Audio Professional Products   TELETRONIX INFORMATION SYSTEMS

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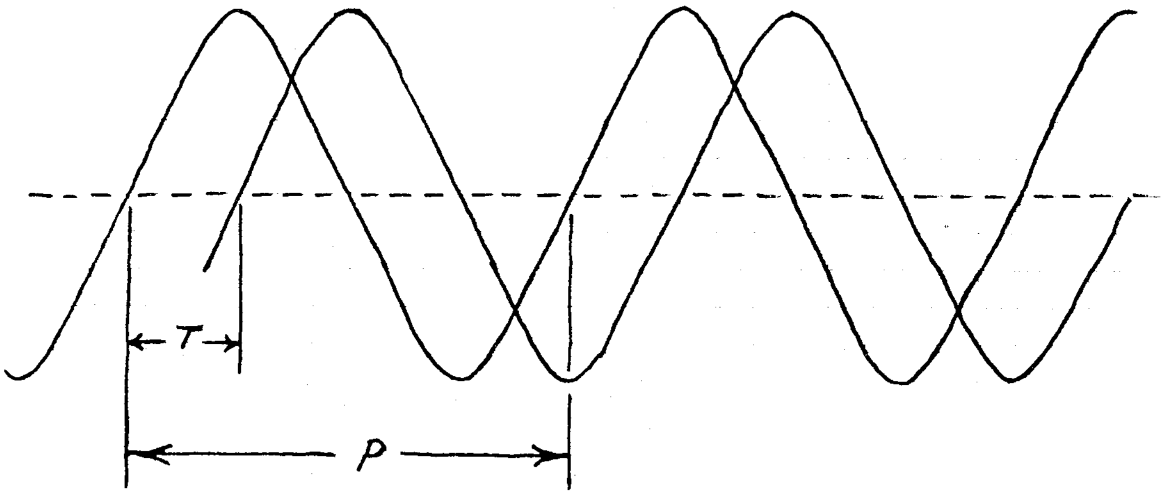
DEFINITION OF RELATIVE PHASE AND PHASE ANGLE



Phase $= \frac{\text{Time interval between zero crossings (T)}}{\text{Wave period (P)}}$ Phase angle $= 360^*\left(\frac{T}{P}\right)$ degrees $= \phi$

EXAMPLE

Phase $= \frac{0.25}{1.0}$ Phase angle $\phi = 360(0.25) = 90^{\circ}$

*Substitution of 2π for 360 results in the phase angle in radians instead of degrees. ♦

DEFINITION OF ABSOLUTE PHASE AND PHASE ANGLE

$t = \frac{1}{f}$ D = Distance in ft or meters c = Velocity of media in ft/sec. or M/sec.

f = Frequency in Hz t = Time period/360°

Absolute phase $= \left(\frac{D}{tc}\right)$ Absolute phase angle $= 360\left(\frac{D}{tc}\right)$

EXAMPLE

A microphone is 10 ft. from a loudspeaker. What is the absolute phase angle between them for a frequency of 1000 Hz? (c = 1130 ft/sec.)

$$\phi_{ABSOL.} = 360\left(\frac{10}{1/1000\ (1130)}\right) = 3{,}185.84^{\circ}$$

FINDING THE VELOCITY OF THE MEDIA

$$c = \frac{D}{t(\text{Phase}_{ABSOL})}$$

$$\text{Phase}_{ABSOL} = \left[\left(\begin{matrix}\text{Time from source to}\\ \text{receiver in secs.}\end{matrix}\right) + \left(\frac{\text{Relative phase angle}}{360}\right)\right]\left[\begin{matrix}\text{Frequency}\\ \text{in Hz}\end{matrix}\right]$$ ♦

S.I. REBELS

We are reliably informed that in England beer is still sold by the pint rather than the liter and tape recorders still operate in inches per second and fractional inch widths. While the schools continue to propagandize for S.I., they'll never reach the kids. Now if TV were to take on the task...... ♦

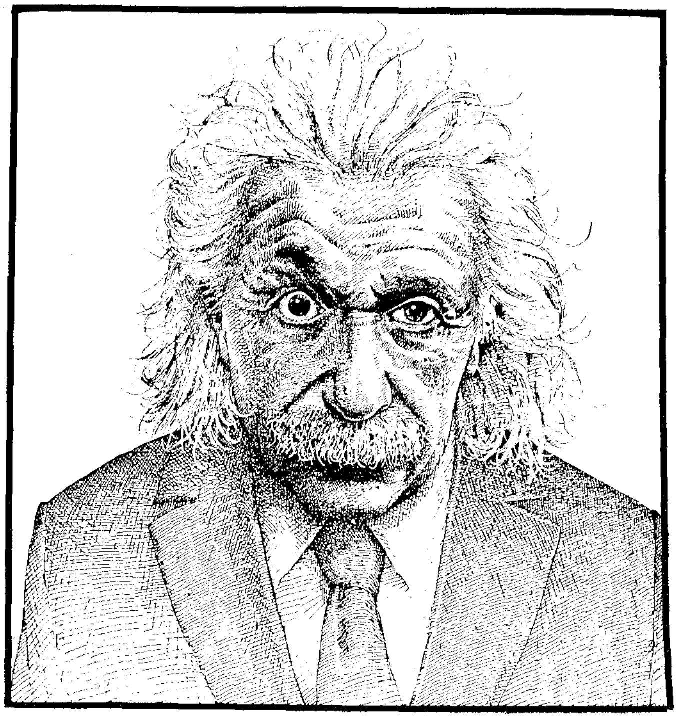
"FACEDNESS" AND ITS RELATION TO ARTISTIC TALENT

On further inquiry into Karl V. Smith's report on his 32 years of study of facial movement given in the *ASA Journal* of June 1984 we quote:

> *"In right faced persons,*
>
> *(1) The right side of the face is more open - i.e., less compressed or drawn up between brow and jaw - than the left side.*
>
> *(2) The right brow is elevated higher than the left brow.*
>
> *(3) Persistent dimples and wrinkles are less marked on the right than on the left side.*
>
> *(4) The head and face are turned and tilted toward the left so that the right side of the face is displayed more in talking, and*
>
> *(5) The right side of the mouth opens farther during speech than the left side."*

In left-faced persons, these five characteristics are reversed and the left side of the mouth and face is relatively more active in articulating speech sounds and singing.

Musically talented, artistic invididuals are left faced. This view of Einstein reveals an extreme right faced individual. ♦



Note about picture from a computer magazine: The Newsletter is being written in Gettysburg while we are traveling. Therefore we don't have our collection of Einstein books with us to verify the accuracy of this computerized photograph.

HANDLING SIGNAL LEVEL CALCULATIONS

WHEN A TERMINATION RESISTOR IS PRESENT

In calculating the effect of a termination resistor on the gain or loss of a system's components, we need to consider the following:

1. The power absorbed by the input circuitry of an amplifier has *nothing to do* with the measurement.

2. Gain or loss always deals with power but it is the output power of the total system.

What the termination resistor will do is cause a variation in E_{IN} because it changes R_{IN}. Since we often measure E_{IN} and *calculate* E_S, we need to be sure to include the effect of R_T on R_{IN} in our computation of the actual R_{IN} our E_{IN} is being measured across.

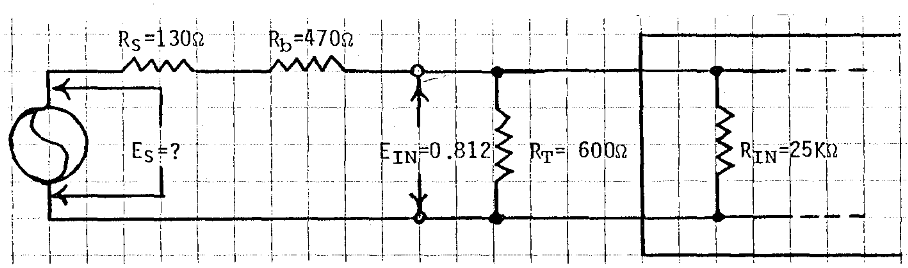
The effective

$$R_{IN} = \frac{1}{\left(\frac{1}{R_T} + \frac{1}{R_{IN}}\right)}$$

Therefore:

$$E_S = E_{IN}\left(\frac{R_{IN(effec)} + R_S}{R_S}\right)$$

L_{AIP} WHEN A TERMINATION RESISTOR IS PRESENT



$$\frac{1}{\left(\frac{1}{600} + \frac{1}{25{,}000}\right)} = 586\Omega$$ (i.e., the effective R_{IN} for L_{AIP} calcu.)

$$130 + 470 = 600\Omega$$ (i.e., the effective R_S for L_{AIP} calcu.)

$$E_S = 0.812\left(\frac{586 + 600}{600}\right) = 1.61\text{v}$$

$$L_{AIP} = \left(\frac{(E_S)^2}{0.001\ R_S}\right) - 6.02\text{ dB} = 10\ \text{Log}\left(\frac{(1.61)^2}{0.001(600)}\right) - 6.02\text{ dB} = 0.31\text{ dBm}$$

In the example case shown, we illustrate a 25 KΩ R_{IN} being shunted by 600Ω with a resulting $R_{IN(effec)}$ of 586Ω. Using this $R_{IN(effec)}$ to calculate E_S, we go from there to the L_{AIP} of 0.31 dBm. ♦

HIGH "FUTILITY" SPEAKER

In a recent class at Griffiss Air Force Base, we encountered a Japanese home high fidelity loudspeaker with some remarkable self cancellation of its own response. This wide band destructive interference was caused by the misalignment of the signal energies from its tweeter and an upper mid range horn (see Fig. 1a). This device had "buttons" on the front labeled "crisp," etc., and pressing one combination resulted in a slightly different response lower in frequency (see Fig. 1b). The buttons, according to the instruction manual, changed levels to the various drivers.

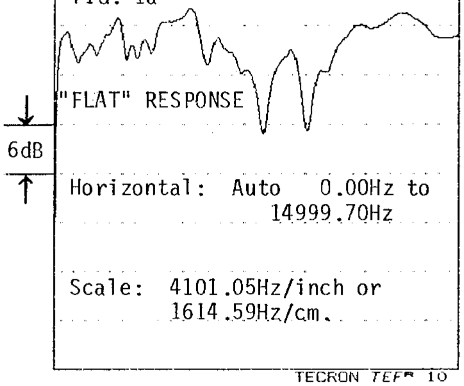


FIG. 1a

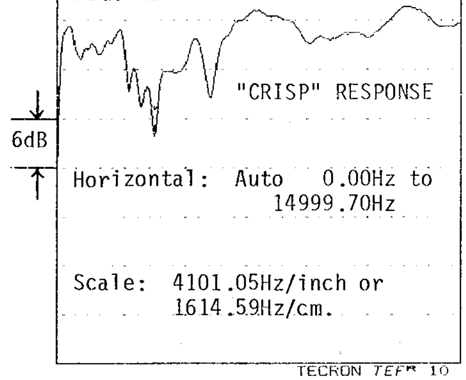


FIG. 1b

While we often read comments in some magazines about high fidelity being in advance of commercial offerings in loudspeaker design, both our ears and our tests so far have said this is often not so. We believe that a very significant area for psychoacoustic research would be the testing of individuals as to their preference between a "live" instrument and a loudspeaker (with both behind a screen at the end of a room) where say a dozen of the most popular hi fi speakers are A - B'ed with an occasional insertion of the "live" instrument, preferably a single instrument such as a violin or piano. We believe, as a hypothesis, that many of today's listeners would not enjoy the "live" instrument. ♦

SYN-AUD-CON BASIC SYSTEM DEFINITIONS

I. Level

There are two defined levels:

1. $$L_{AIP} = 10 \text{ Log}\left(\frac{(E_S)^2}{0.001R_S}\right) - 6.02 \text{ dB}$$

2. $$L_{OUT} = 10 \text{ Log}\left(\frac{(E_L)^2}{0.001R_L}\right)$$

L_{AIP} is the available input power level in dBm for all devices within a system that interconnect to other devices.

L_{OUT} is the actual dissipated power level in dBm at the output of the system.

The word "level" should be limited to quantities in decibels.

II. Level Difference

The *difference* in level may be observed by any voltage, current, sound pressure or similar measuring instrument across a fixed resistance point by using

$$L_{diff} = 20 \text{ Log}\left(\frac{\text{Quan } 1}{\text{Quan } 2}\right)$$

Note that Quan 1 and Quan 2 *are not levels*.

III. Gain and Loss

When used in systems design work, "gain and loss" refer to *insertion* gain or loss.

$$G = (L_{OUT} - L_{AIP})$$

$$G = (L_{AIP} \text{ of device furthest along} - L_{AIP} \text{ of earlier device})$$

$$G_{system} = 20 \text{ Log } \frac{E_{OUT}}{E_{IN}} + 20 \text{ Log } \frac{R_{IN}}{R_S + R_{IN}} + 10 \text{ Log } \frac{R_S}{R_L} + 6.02 \text{ dB}$$

Notes

There are no limitations on what R_{IN}, R_S and R_L values are.

The L_{AIP} is entirely independent of the R_{IN} of the device it is connected to.

Addition of insertion gains and losses yield the same G_{system} as the G_{system} equation. ♦

THE DIFFERENCE BETWEEN
VOLTAGE AMPLITUDE AND SIGNAL LEVEL

Those unfamiliar with audio *systems* often question the fact that you can't measure *level* with a voltage measuring instrument (often mislabeled a volt*meter*). The following is a specific example.

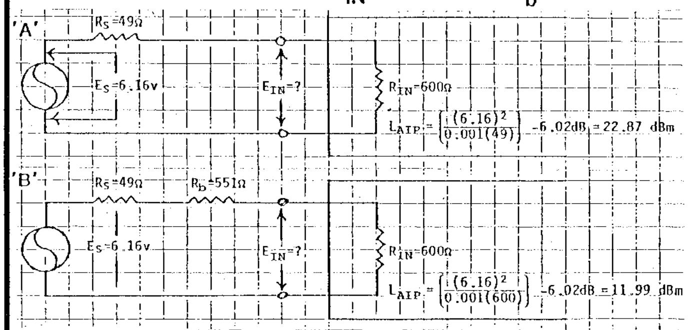
Case 'A'

Case 'A' is a circuit that has a test oscillator connected to an audio amplifier. The E_S of the oscillator is 6.16v. The R_S of the oscillator is 49Ω. The R_{IN} of the amplifier is 600Ω; therefore, the E_{IN} developed by the oscillator at R_{IN} is 5.69v. The L_{AIP} of the oscillator is 22.87 dBm. The output of the amplifier is +50 dBm. Therefore, by definition the *gain* of the amplifier is: +50 - 22.87 = 27.13 dB.

Case 'B'

We now add a buildout resistor of 551Ω to the output of the oscillator in order to make it appear as a 600Ω source. Our R_S then becomes 49 + 551 = 600Ω. Our L_{AIP}, due to the impedance change, becomes 11.99 dBm. This means that the *level* at this point in the circuit has changed by 22.87 dBm - 11.99 dBm = 10.88 dB. What is the change in level at the output of the system? To determine that, we need to recognize that the addition of the "buildout" resistor not only changed the L_{AIP} but the gain of the amplifier as well. The new E_{IN} is 3.08v. Therefore, the amplifier input is:

CHANGES IN E_{IN} CAUSED BY R_b



$$E_{IN} = \left(\frac{E_S\ R_{IN}}{R_S + R_b + R_{IN}}\right) = \left(\frac{6.16\ (600)}{49 + 600}\right) = 5.69\text{v}$$

$$\left(\frac{6.16\ (600)}{49 + 551 + 600}\right) = 3.08\text{v}$$

$$20 \text{ Log}\left(\frac{3.08\text{v}}{5.69\text{v}}\right) = -5.34 \text{ dB}$$

$$20 \text{ Log}\left(\frac{R_S + (R_S + R_b)}{R_S + R_b + R_{IN}}\right) = 20 \text{ Log}\left(\frac{49 + (49 + 551)}{49 + 551 + 600}\right) = -5.34 \text{ dB}$$

$$20 \text{ Log}\left(\frac{5.69\text{v}}{3.08\text{v}}\right) = -5.33 \text{ dB less than Case 'A'}$$

Therefore, the gain becomes: +50 dBm - 5.33 dB - 11.99 dBm = 32.68 dB.

The difference in gain for Case 'A' and Case 'B' becomes: 32.68 - 27.13 = 5.55 dB.

The difference in input voltages is 5.33 dB: 5.55 + 5.33 = 10.88 dB which is the difference in *level* at the input of the amplifier, 5.55 dB due to the difference in gain of the amplifiers, 5.33 due to difference in input voltage to the amplifiers.

Conclusion: A voltage measuring instrument can read changes in level. It cannot read *levels*. ♦

B&K "TECHNICAL REVIEW"

Brüel and Kjaer's "Technical Reviews" and "Application Notes" are some of the most reliable sources of technical information in audio measurements available today. They are not afraid to call "a spade a spade," even when their own product is the subject under discussion. For example, in their "Technical Review No. 1, 1984," on page 50, they clearly point out the inapplicability of their new dual channel FFT (which, incidentally, we consider the best of the currently available FFTs) to testing "mechanical systems having non-linear spring elements" (i.e., read loudspeakers, microphones, and other transducer forms) and further on, the statement that the study of "harmonic distortion will also require a sinusoidal excitation" followed by a recommendation to look into TDS for such measurements. Such frankness in today's world bespeaks an unusual integrity on their part.

One of their latest "Application Notes" is entitled "The Use of B&K Omnidirectional Microphones for Modern Recording." While the author of the note kept to conventional microphone techniques (but very interestingly so), we merely want to add the comment that reading this note plus knowledge of the Haas distance between microphones for elimination of "leakage" and the use of these microphones in conjunction with a boundary offer a much enhanced technique for the faithful recording of a sound field's spatial geometry while preserving all the encoded nuances of amplitude, signal delay, and relative phase differences from the source being recorded. We highly recommend study of this "Application Note" plus some careful in-the-field experimentation. ♦

NEWSLETTER SUBSCRIPTION INCREASES 6.66%

It has been a long time since we increased the price of our Newsletter subscription. Starting with this issue the subscription rate will be $32.00 per year for graduates in the United States and $38.00 per year for all other countries. We're pleased we were able to hold the increase to a modest 6.66%. This issue, Volume 12, No. 1, starts our twelfth year of publication. This is only the second price increase since we started. Our Newsletter subscription, like our Workshops, are subsidized because we feel so strongly about their worth to the audio industry. We hope you agree. ♦

CONVERTING AMPLIFIER INPUT SENSITIVITIES INTO L_{AIP}

Many manufacturers give amplifier input *sensitivity* rather than input level. The Crown PS-200, for example, states that 1.3 volts at the input (30KΩ) produces 90 watts into 8Ω at each channel.

$$10\ \text{LOG}\left(\frac{90\text{w}}{0.001\text{w}}\right) = 49.54\ \text{dBm}$$

The question then becomes what L_{AIP} will drive this amplifier to full output. In order to find the L_{AIP} we need the R_S of the device that will be connected to this amplifier. Let's say we're going to use a Shure 267 mixer. Its R_S = 130Ω

$$L_{AIP} = 10\ \text{LOG}\left(\frac{\left(\frac{E_{IN}\ (R_S + R_{IN})}{R_{IN}}\right)^2}{0.001\ R_S}\right) - 6.02\ \text{dB} \qquad L_{AIP} = 10\ \text{LOG}\left(\frac{\left(\frac{1.3\ (130 + 30{,}000)}{30{,}000}\right)^2}{0.001\ (130)}\right) - 6.02\ \text{dB} = +\ 5.16\ \text{dBm}$$

The *Gain* of the amplifier is then: 49.54 dBm - 5.16 dBm = 44.39 dB.

Amplifier Level And Gain Specifications

We can now write: Input level required for full output + 5.16 dBm   Output level +49.54 dBm   Gain 44.39 dB and we can use these values to properly label a *system* gain chart when these components are employed in that system.

Gain overlap between the Shure mixer and Crown amplifier is +18 dBm - (+5.16 dBm) = 12.8 dB. If we desire a 6 dB headroom margin and allow 10 dB for meter lag, then we'll operate the Shure mixer at 18 dBm - 10 dB - 6 dB = + 2VU. Therefore, with the sensitivity control on the Shure mixer set at + 4VU we would not allow any level to exceed an instrument indication (i.e., scale reading) of -2VU. We would have 12 dB of "working loss" available in the controls and would not require a "pad" between these two units. If a passive equalizer were to be used between them, the gain overlap is marginal. ♦

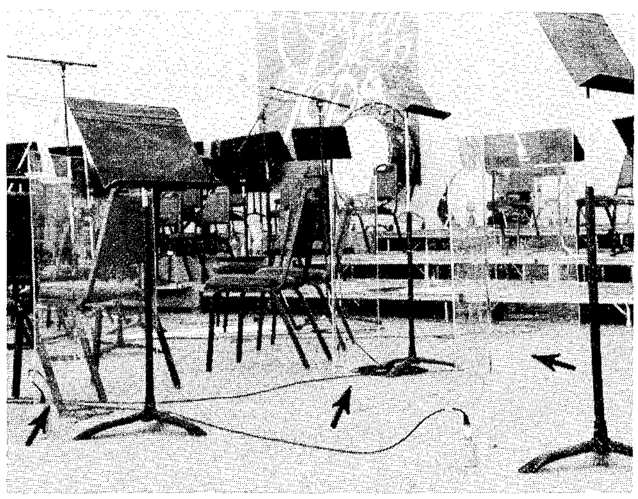
ARTISTRY AND TECHNOLOGY

Dave Johnson adjusting the PZM "dishes" for the outdoor concerts of the San Diego Symphony.

When engineers like Dave Johnson and innovators like Ken Wahrenbrock vector together, artistry and technology benefit. It has been said that the future of the PZM lies in the innovative use of the PZM and aycrilic.

Here's some more evidence of what can be done. These microphones are used with the San Diego Symphony. ♦

Note the size, placement and shape of the music stands and you'll begin to build a growing appreciation of their role in obtaining orchestral diffusion.



Ken made 4' x 18" per side PZMs for the string basses.

CONVERTING AN AMPLIFIER'S SENSITIVITY RATING

INTO ITS GAIN SPECIFICATION

Most contemporary power amplifiers have a high impedance input, i.e., R_{IN} - 47,000Ω, and a sensitivity rating of "x volts in yields y watts out", i.e., 0.5v for full power (200w).

That figure 0.5v is an E_{IN} value. The 47,000Ω is an R_{IN} value. The 200 watt output is for a load rating of 4, 8, 16Ω, or some value. Let's for the sake of an example say it is 8Ω.

Output Power Level

The output power *level* in dBm equals:

$$L_{OUT} = 10 \text{ LOG} \left(\frac{P_{OUT}}{0.001w}\right) = 10 \text{ LOG} \left(\frac{200w}{0.001w}\right) = 53.01 \text{ dBm}$$

Input Power Level

Because the power drawn from the source or the power dissipated at the input of the power amplifier does not enter into the calculation of the power amplifier's gain, the figure we are after is the L_{AIP} rating for some source device likely to be connected to this amplifier. Therefore, we ask the question this way. If I choose to use a Shure mixer (R_S = 150Ω), what L_{AIP} must it deliver at the input of my amplifier in order to generate an E_{IN} = 0.5v?

$$L_{AIP} = 10 \text{ LOG} \left(\frac{\left(\frac{E_{IN}\ (R_S + R_{IN})}{R_{IN}}\right)^2}{0.001\ R_S}\right) - 6.02 \text{ dB} = 10 \text{ LOG}\left(\frac{\left(\frac{0.5\ (150 + 47{,}000)}{47{,}000}\right)^2}{0.001(150)}\right) - 6.02 \text{ dB} = -3.77 \text{ dBm}$$

Amplifier Gain

This then allows the *Gain* of this power amplifier when used with this mixer to be calculated

$$G_{AMP} = (+53.01) - (-3.77) = 56.78 \text{ dB}$$

Gain Overlap

Since the specification on the Shure mixer says it has a full power output of +18 dBm, this provides us with (+18) - (-3.77) = 21.77 dB of *gain overlap*. Because we will allow 10 dB of instrument lag and 6 dB of headroom, this actually leaves us with (+18) - (+16) - (-3.77) = 5.77 dB of gain overlap, i.e., we'll run the mixer at almost full gain in order to fully drive the power amplifier.

Output Voltage Of Power Amplifier

Finally, what E_{OUT} should we measure when the power amplifier is actually putting out 200 watts from a 0.5 volt input?

$$E_{OUT} = \sqrt{(0.001\ (R_L))\left(10^{\left(\frac{L_{OUT}}{10}\right)}\right)} = \sqrt{(0.001(8))\left(10^{\left(\frac{+50}{10}\right)}\right)} = 28.28v$$

It's of interest to note that the voltage amplifications between input and output voltages yields:

$$20 \text{ LOG} \left(\frac{28.28v}{0.5v}\right) = 35.05 \text{ dB}$$

whereas the actual *gain* is 56.78 dB.

Summary

Because technicians normally use voltmeters and simple direct reading impedance meters, this technique allows full use of these basic tools to obtain via easy calculation true levels, gains or losses, and gain overlaps.♦

SALES VS. SERVICE

A series of fascinating newspaper articles about the Sony Corporation have arrived from various sources in the past few weeks.

During the Olympic games, Sony and ABC were allowed to use 500 milliwatts instead of the normal 30 milliwatts for their wireless microphones and were allotted a special temporary authority to employ a currently unused UHF channel for their transmissions. The press releases tell us that the Sony wireless microphones worked perfectly without interference at the Olympics. The press releases rarely tell you why.

The second writeup was in the Chicago Tribune and documented Sony's cavalier attitude toward servicing their products. Six months was not untypical, it was reported.

Syn-Aud-Con's own observations of the Sony digital recording push is replete with disregard for the pollution of historical recorded events, so long as the pollution is done to their specifications. Such behavior is not Japanese, as we have also witnessed exceptionally ethical practices by Japanese manufacturers in many fields. ♦

FINDING E_{IN} FOR A GIVEN L_{AIP}

When you, as the systems engineer, have drawn up a *gain* chart for a system, you often need to specify for the installer just what E_{IN} he should read with his voltmeter when he connects two components together.

$$E_{IN} = \frac{\sqrt{(0.001(R_S))\left(10^{\left(\frac{L_{AIP} + 6.02dB}{10}\right)}\right)R_{IN}}}{(R_S + R_{IN})}$$

Example

You want to operate a mixer at a maximum level of +8 dBm. The mixer has an R_S of 600Ω . The amplifier it drives has an R_{IN} of 30 KΩ. What E_{IN} across R_{IN} represents an L_{AIP} of +8 dBm?

$$E_{IN} = \frac{\sqrt{(0.001(600))\left(10^{\left(\frac{(+8) + (+6.02)}{10}\right)}\right)(30{,}000)}}{(600 + 30{,}000)} = 3.81 \text{ volts}$$

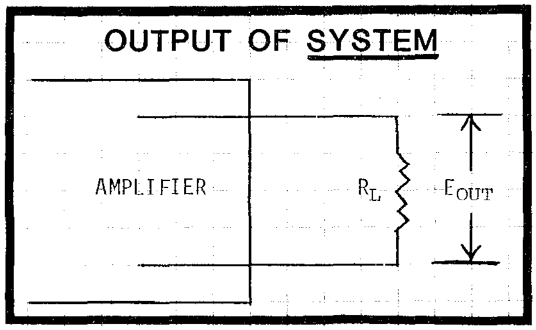
Basic Rules

1. At the output of every "part" (component) of a system *except the final output device*, we use L_{AIP} for levels and E_{IN} to the next device it will drive.
2. At the final electrical output (the power amplifier output), we calculate and measure actual power in the load.

$$L_{OUT} = 10 \text{ Log}\left(\frac{(E_{OUT})^2}{0.001 \; R_L}\right)$$

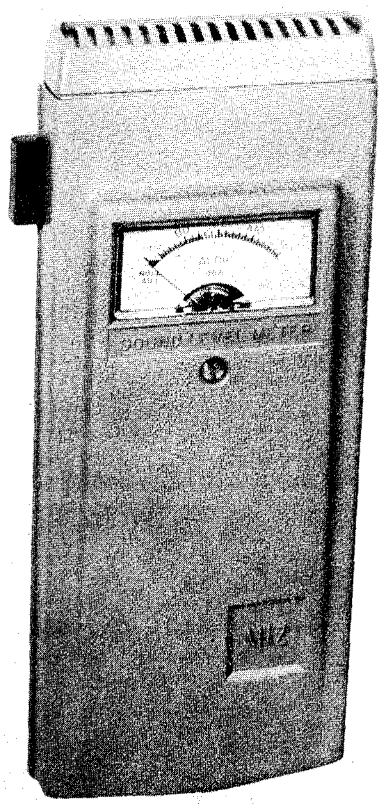
and:

$$E_{OUT} = \sqrt{(0.001(R_L))\left(10^{\left(\frac{L_{OUT}}{10}\right)}\right)}$$



A definition of L_{OUT} is shown above. ♦

LOW COST SOUND LEVEL METER



It is not easy to find a low cost sound level meter. S/Sgt. Sherry Ross at Camp Pendleton Marine Base (Anaheim class, 1984) sent us literature on VIZ test equipment. We ordered one to try it out and we think it is worth the $139.00 asking price for measuring excessive noise.

To order their test equipment catalog, write:

VIZ Manufacturing Company
335 East Price Street
Philadelphia, PA 19144-5782

Telephone: (215) 844-2626
or
(800) 523-3696 ♦

NEW TELEPHONE NUMBER

After November 3, 1984,
Syn-Aud-Con's new telephone number will be
(714)728-0245.

HOW TO DOUBLE CHECK THE GAIN STRUCTURE OF A SOUND SYSTEM

Let's suppose for the moment that you have never used a Shure SM81 with a Shure M267 mixer with a White 4004 passive equalizer driving a Crown D-75 power amplifier, which in turn is connected to a Rauland MLS-3A two-way loudspeaker system. Your job is to look over the specification sheets and make sure that these drivers have the necessary *gain overlaps* to be able to work together.

What Is Gain Overlap?

Gain overlap is the amount of excess level available at the input of the driven device above that needed to drive it to full output.

Looking at the Crown D-75 spec, we see that its input sensitivity is 0.812 volts for a full output. In our example system, since the Rauland MLS-3A is an 8 ohm loudspeaker, full output is 35 watts into 8 ohms. The input impedance is 25KΩ. Because the White 4004 is a passive unit, it must operate into a 600Ω load. This means that the preceding device (the 4004) must develop 0.812 volts across 25KΩ shunted by 615Ω

$$\frac{1}{600} - \frac{1}{25{,}000} = \frac{1}{615\Omega}$$

Thus, the R_{IN} of the D-75 looks like 600Ω to the 4004.

The E_S that must be present then becomes:

$$E_S = E_{IN}\left(\frac{R_S + R_{IN}}{R_{IN}}\right) \qquad E_S = 0.812\left(\frac{615 + 600}{600}\right) = 1.64 \text{ volts}$$

$$L_{AIP} = 10 \text{ LOG}\left(\frac{(E_S)^2}{0.001\ R_S}\right) - 6.02 \text{ dB} \qquad L_{AIP} = 10 \text{ LOG}\left(\frac{(1.64)^2}{0.001(600)}\right) - 6.02 = 0.52 \text{ dBm}$$

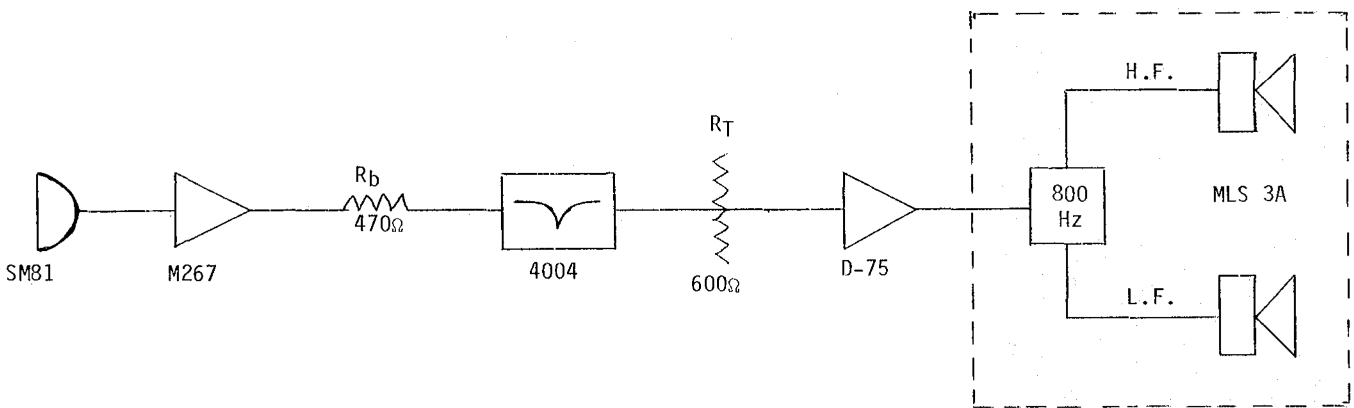
The power output level of the amplifier is: $L_{PWR\ OUT} = 10 \text{ LOG}\left(\frac{35w}{0.001w}\right) = 45.44 \text{ dBm}$

Therefore, the *Gain* of the D-75 used in this configuration is: 45.44 dBm - 0.52 dBm = 44.9 dB. The maximum output level of the Shure M267 mixer is + 18 dBm. The Gain overlap is: +18 dBm - 0.52 dBm = 17.9 dB.

In the attached illustration we show a user that requires the Shure M267 mixer to have a "working gain" of 78.8 dB. The White 4004 is found to have a 10 dB *loss* after equalization is done. The Crown D-75 thus uses a working gain of 37.4 dB

78.8 - 10 + 37.4 = 106.2 dB of electrical gain.

A Simple Sound System Set Up



SPECIFICATIONS GIVEN

SM81	sensi. = 0.63 mv/UBAR Z = 150Ω
M267	+ 18 dBm output into 600Ω Actual Z_{OUT} = 130Ω
4004	Assume total loss of 10 dB
D-75	Input Z = 25KΩ Output Z Rating = 8Ω Output power 35w Sensi = 0.812v for full output
MLS 3A	97 dB at 4' from 1 watt input

SAME SPECIFICATIONS IN A MORE USEFUL FORMAT

SM81	EIA sensi = -135.8 dBm
M267	L_{AIP} = 18.6 dBm input Z = 800Ω Z_{OUT} = 130Ω
4004	Z_{IN} = 600Ω Z_{OUT} = 600Ω Loss = 10 dB
D-75	Output power level = + 45.4 dBm Gain (600Ω Source) = 45 dB
MLS 3A	EIA Sensi = 49.5 dB

Continued....

SYSTEM LEVELS

If we have a talker that produces an Lp = 75 dB at the microphone, then:

Input level at mixer = -135.8 + 75 = - 60.8 dBm

Output level of mixer = + 18 dBm

Output level of 4004 = + 8 dBm

Output level of D-75 + + 45.4 dBm

Level at listener 30' in front of MLS-3A = Lp = + 45.4 dBm + 49.5 dB = 94.9 dB

Working gain of M267 = ((+ 18) - (- 60.8)) = 78.8 dB

Working gain of D-75 = (+45.4) - (+8) = 37.4 dB

Gain overlap between M267 and D-75 = (18 dBm) - (0.31 dBm) = 17.7 dB

(78.8) - (10) + (37.4) = 106.2 dB Electrical Gain

A Quick Check

An extremely useful formula for gain of a system is:

$$G = 20\ \text{LOG}\left(\frac{E_{OUT}}{E_{IN}}\right) + 20\ \text{LOG}\left(\frac{R_S}{R_S + R_{IN}}\right) + 10\ \text{LOG}\left(\frac{R_S}{R_L}\right) + 6.02\ \text{dB}$$

$$E_{IN} = \frac{E_S \quad R_{IN}}{R_S + R_{IN}} = \left(\frac{0.00071\ (800)}{150 + 800}\right) = 0.00060$$

In our system's case:

$$20\ \text{LOG}\left(\frac{E_{OUT}}{E_{IN}}\right) = \frac{16.73}{20\ \text{LOG}}\left(\frac{16.73}{0.00064}\right) = 88.91\ \text{dB}$$

$$20\ \text{LOG}\left(\frac{R_{IN}}{R_S + R_{IN}}\right) = 20\ \text{LOG}\left(\frac{800}{150 + 800}\right) = -1.49\ \text{dB}$$

$$10\ \text{LOG}\left(\frac{R_S}{R_L}\right) = 10\ \text{LOG}\left(\frac{150}{8}\right) = 12.73\ \text{dB}$$

$$88.91 - 1.49 + 12.73 + 6.02 = 106.2\ \text{dB}$$

What Do You Do Next?

If you're smart, you file away all this hardwon data in a master book of components you use in sound systems so that you don't have to do them again for this set of circumstances. This should be done especially for the EIA sensitivity figures of the microphones and loudspeakers you use, the output power levels in dBm for the power amplifiers in your inventory, and a complete listing of all impedance ratings accompanied by the actual *measured* ratings. Once all of this is done *system design* work becomes largely addition and subtraction.♦

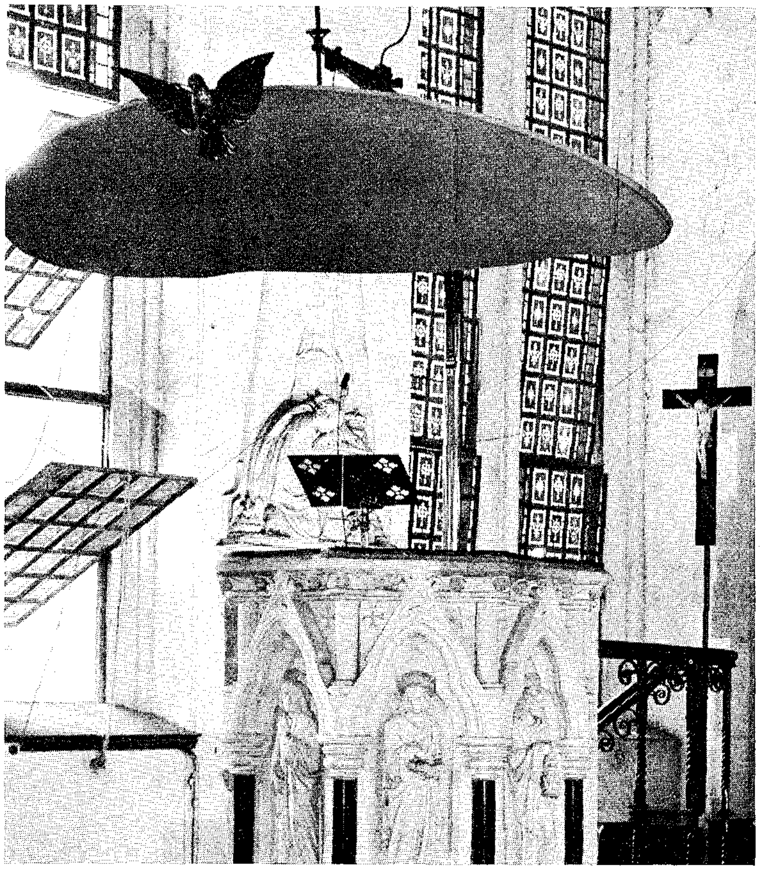
NEW ADVENTURES

Henry Martin - Formerly with Harrison Systems, Nashville, Tennessee, is now Senior Engineer with JBL working with Professional Series Products. ♦

SMILE

Bruce Thayer, Vice-President of network engineering for Teleconnect™ sent us the following from a state commerce commission hearing when the telephone engineer described telephone ringing as *"alternating two second bursts of energy and four second bursts of silence."*

Bruce went on to comment, "*No doubt this is done by incorporating the silence generator I reported to you several years ago (essentially two terminals on the end of a tomato can connected together with #10 solid copper wire).*" ♦



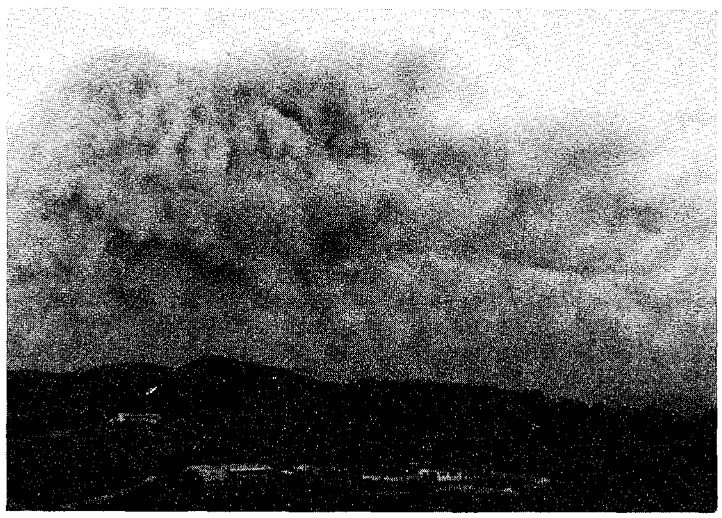
PULPIT CANOPY

The photograph shown here was sent to us by Peter R. Ames, Director of Sports and Games Ltd., Port of Spain, Trinidad. The reflection over the pulpit has adjustments for both height and pitch.

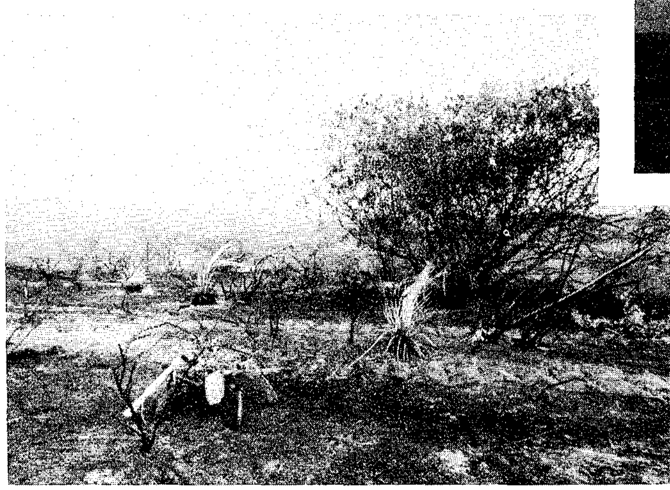
We are told the pulpit was built in 1902. Reflections such as this not only return useful energy from the talker to the audience area, but can serve as a shield between the microphone and the loudspeaker in some cases. Care must, of course, be taken to insure that the comb filters thus generated fall within a non-destructive interference zone (usually anything with a path length difference in excess of 3 feet). ♦

HABITATION HAZARDS

To those Syn-Aud-Con graduates that made the hike up the hill in back of the seminar center here in California, this view from the top towards the southeast should inspire a certain awe. This is the smoke from one of our forest fires as it raced first toward and then, thank goodness, past Rancho Carrillo.



The second photograph is taken down near the gate on the Ortega Highway and shows what happened to the high grazing grass, cactus groves, and chaparral. In fact, the first mile up our private road from the gate resembled an excellent likeness of a moonscape.



The large cloud fire above was set by a malcontent during an 80 mph wind condition known locally as a "Santa Ana." He or she (undetermined as they have not been caught) drove along hurling lighted flares into the tinder-dry brush.

The fire that devastated the gate area was courtesy of the EPA when a car ran off the Ortega Highway to avoid a head-on collision and its catalytic converter set a fast moving fire that burned hundreds of acres before 5 bombers got it stopped.

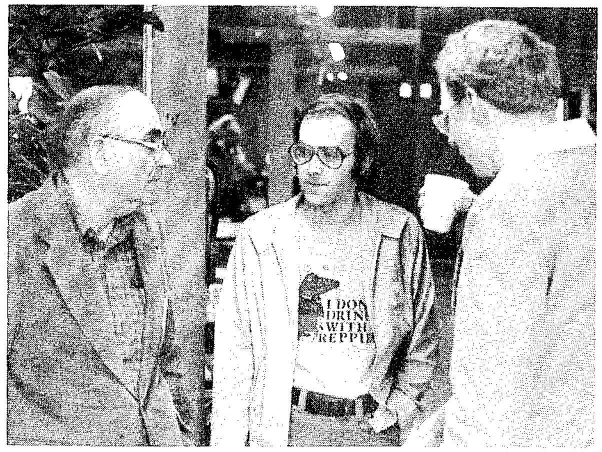
We maintain that every *cloud* has a silver lining, and in this case, we do have more extensive than usual "firebreaks" on two sides of Rancho Carrillo. ♦

USEFUL SENSITIVITY CHART

Bob Hagenbach of Indianapolis has generated a useful chart for the quick conversion of one loudspeaker sensitivity rating to another. Remember that a loudspeaker's sensitivity *is not* its efficiency. (See the loudspeaker efficiency equation in Newsletter Volume 11, No. 2, page 27.) The conversion factor for a 1.0 meter rating into a 0.283 meter rating is:

$$20 \text{ Log}\left(\frac{1}{0.283}\right) = 10.96 \text{ dB}$$

Taking the *1 watt* at 0.283 meter sensitivity rating and subtracting 10 Log Q from it gives the L_W for a 1 watt electrical input. Using Scales 5 and 10 on the Syn-Aud-Con slide rule then allows you to subtract from +50 on Scale 10 (100% on Scale 5) the sum of 120 minus the L_W which is the value in dB that the loudspeaker is *below* 100%.



Bob Hagenbach (center) shown talking with Bob Reid of JoyBob Associates, Monrovia, California, (left) and John Prohs of Ambassador College during a Loudspeaker Array Design Workshop at Rancho Carrillo.

Example

A loudspeaker is 99 dB at 4 ft. with a Q = 7, its 1 meter rating is:

$$99 + 1.72 = 100.72 \text{ dB}$$

$$100.72 - 10 \text{ Log}(7) = 92.27 \text{ dB}$$

Converting this value to 0.283m, we obtain

$$92.27 + 10.96 = 103.23$$

which is the L_W of this loudspeaker. $120 - 103.23 = 16.77$ dB Therefore, we have a loudspeaker with an efficiency 16.77 dB below 100%.

The power ratio of -16.77 dB times 100 equals the percent efficiency: $10^{\left(\frac{-16.77}{10}\right)} \times 100 = 2.11\%$ ♦

SPL changes of commonly used distances used for horn-driver sensitivities.

A \ B	3 ft.	4 ft.	10 ft.	30 ft.	1M	3M	10 M
3 ft.	0	-2.5	-10.45	-20	-0.78	-10.32	-20.78
4 ft.	2.5	0	-7.96	-17.5	1.72	-7.82	-18.28
10 ft.	10.45	7.96	0	-9.54	9.68	1.02	10.32
30 ft.	20	17.50	9.54	0	19.22	9.68	0.78
1 M	0.78	-1.72	-9.68	-19.22	0	-9.54	-20
3 M	10.32	7.82	-0.14	-9.68	9.54	0	-10.46
10 M	20.78	18.28	10.32	0.78	20	10.46	0

EXAMPLE:

Which horn-driver is more sensitive? (Both are measured with DI = 10 and 1W)

#1 SPL at 4 ft = 115 dB
#2 SPL at 3M = 109 dB

To find SPL change from 4 ft to 3M let A = 4 ft and B = 3M then change = -7.82 dB

#1 SPL at 3M = 115 -7.82 = 107.18 dB
#2 SPL at 3M = 109 - 0 = 109 dB

Therefore #2 is more sensitive.

$$\text{dB change} = 20 \log\frac{A}{B}$$

CLASSIFIED

FOR SALE: Alan Lubell of Underwater Loudspeaker fame has 75 transformers (#51-120) he would like to sell. He wants $10 each, plus shipping. They are biphilar wound, hence have an excellent center tap. He says he made up a Grutzmacher bridge using the transformer and "it worked very well."

Contact: Alan Lubell, 21 N. Stanwood Rd., Columbus, OH 43209.
Telephone: (614) 235-6740

WANTED: 2 part-Catalytic Polyurethane Foam for treatment of 1/4" curved-horn - Flare walls in LF horns (etc.).

Contact: Tom Young, Electro-Acoustic Resources, P.O. Box 436, Naugatuck, CT 06770. Telephone: (203) 729-5368

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Syn-Aud-Con receives tangible support from the audio industry, and twelve manufacturing firms presently help underwrite the expense of providing sound engineering seminars. Such support makes it possible to provide the very latest in audio technology while maintaining reasonable prices relative to today's economy and to provide all the materials and continuing support to all graduates of Syn-Aud-Con.

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. They are "in-tune" with what a Syn-Aud-Con graduate needs.

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