SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

newsletter

P.O. Box 669, San Juan Capistrano, CA 92693
Ph: 714-728-0245

Volume 12, No. 2
Winter 1985

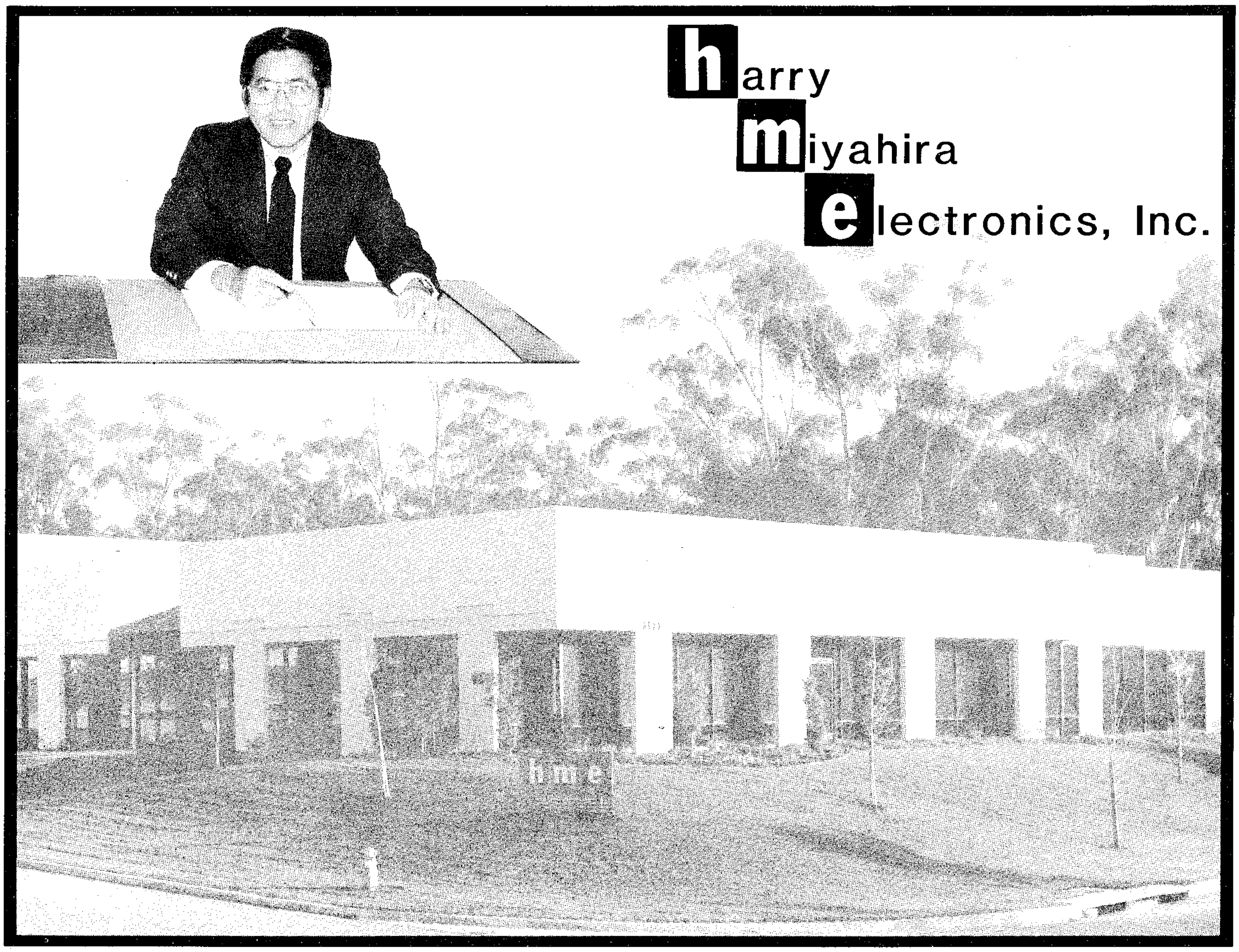
SYNERGETIC
Working together; co-operating, co-operative

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

EXCHANGE OF IDEAS

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

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



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HM ELECTRONICS, INC.

Harry Miyahira was born in Hawaii, graduated from Purdue University, worked for the lengendary Ralph Townsley at WBAA, and founded HME in 1971. Harry picked a tough business but proved his worth by becoming the top company in the highly competitive wireless microphone business.

The first radio microphone developed and manufactured by HME was sold under the Electro Voice label. Since that time, HME has expanded the product line and offers a variety of wireless microphones to suit professional application needs as well as industrial and institutional requirements.

HME engineers also developed a full line of wireless and cabled intercom equipment for the entertainment and industrial markets. Interface units allow combining both wireless and cabled systems.

Working with Syn-Aud-Con, HME introduced the PAL System (Precision Audio Link) which permits audio and acoustic engineers to make precision measurements via a wireless transmission link.

HME recently moved into its 20,000 sq. ft. sales, engineering and administrative headquarters in Scripps Ranch Industrial Park. Combined with its Mexico operations, HME continues to grow as a major manufacturer in the wireless microphone and communications industry.

We have used HME's latest wireless microphone all over the United States and it has been completely interference free. It has been a pleasure to use. ♦

Harry Miyahira and his staff at HM Electronics, Inc.

STAFF

Editors:
Don & Carolyn Davis

Design & Production:
Jan Kreitz
Pat Carlson

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EDITORIAL - THE GOLDEN AGE OF AUDIO?

Audio, today, is at the threshold of remarkable advances in the quality of sound that we will hear at public events. The past decade was an overpowering one of more and more. This decade will be one of better and better. We are just learning the secrets of controlling and distributing the power we have available in complex arrays. We are seeing for the first time how devices with beautiful polar responses can degenerate into totally out-of-control polar responses when combined in arrays. We now see how the transient response of electronic devices such as power amplifiers can be remarkably distorted by everyday real-life loads the designers just don't consider.

TEF® Analysis

TEF analysis is far exceeding even our fondest hopes for it. It's hard to realize that in the midst of a world mesmerized by an older outdated technology the Heyser transform is going to totally prevail. Fourier analysis is backed by the authority of dogmatic academicians, manufacturers, engineers, and end users unaware of the powerful Heyser transform at hand. Why? The Heyser transform is a fundamental improvement in the understanding of how energy is *and can be* utilized in the real world.

Advances That Freed

The greatest enjoyment available in this world is the quest for knowledge supported by energetic learning that flowers into the understanding called wisdom. True science is embodied in the mental state that is equipped to ask the right questions, accept humbly the answers that are true but not the ones expected, and the sharing with others that simultaneously submit the claim to proof and give it life.

Freedom has come to mankind in many guises. The firearm made the downtrodden peasant the physical equal of the armored knight on horseback. The printing press made the genius of the common man equal to the esoteric knowledge of the priest craft. The computer is leveling the difference between academia and real world craftsman. Syn-Aud-Con's perpetual goal is the sufficient distribution of basic audio facts so the recipient of them is free of any type of audio guru.

1985 - The Golden Age of Audio?

The 1980s will be looked back on in audio history as particularly volatile years of fundamental change. 1985 could easily be picked as an example of audio's second "Golden Age." To enjoy the moment to its fullest we will need to seek both the artistry and the science of our chosen field of endeavor. Remember, products will pass; ideas have lives of their own that we are privileged to share in, assist to grow, and send into the world like promising children. ♦

NEW ANALOG SPEECH PROCESSOR

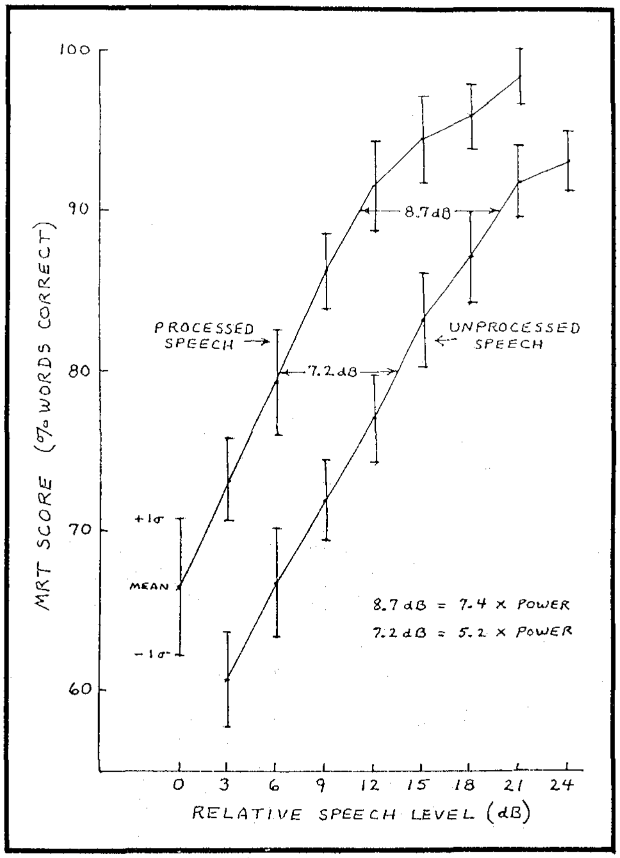
We are reproducing Craig R. Allen's memo on his analog speech processor. Craig is in the Speech Systems and Interference Branch (which is headed by Elaine Schiller) of the Naval Ocean Science Center in San Diego. This unit has to be heard to be believed. Craig is conservative in his description of its performance. It is truly startling to first hear *total* unintelligibility (so total you're not even sure there is speech signal present in the noise you are hearing) followed by, to my ears, total intelligibility when the special processor is switched in *without any change* in the total system power. When the speech signal is raised 10 dB (ten times the power) over the ambient noise, the intelligibility improves but doesn't seem, to my ears, to have the same clear quality the processor provides at 1/10 the power.

This processor works by heterodyning the voice signal to R.F., severely clipping it, and then low-passing it and returning to the voice frequencies. There are, of course, a great deal of additional subtleties but this gives you the basic idea behind this extremely effective technique.

It is Syn-Aud-Con's belief that this processor could literally revolutionize high power paging in noisy areas such as automobile racing facilities, airports, large industrial plants, and large stadiums and arenas.

IMPROVING ANNOUNCING SYSTEM INTELLIGIBILITY
BY ANALOG SPEECH PROCESSING

NOSC Code 441 has developed a new preamplifier for shipboard announcing systems that uses analog processing of the voice signal to enhance intelligibility in noisy spaces. Informal demonstrations using a standard test paragraph and a variety of listeners have shown that processing can make the difference between barely detecting a totally unintelligible voice and understanding nearly every word. Most listeners judged that quadrupling the unprocessed speech power was slightly less effective than inserting the processor. These subjective results have now been verified and made quantitative by formal intelligibility tests, summarized on the graph at right.



Intelligibility was measured by the modified Rhyme Test (MRT) using two talkers, five listeners, and six different lists of 50 words each. Each data point is an average of 15 individual scores. Mean and standard deviation for each point are shown on the graph to indicate the spread of data. An 80% MRT score is considered the minimum acceptable intelligibility for most military communication tasks; 95% of standard test sentences will be understood at this level. A 90% MRT score indicates a high-quality system.

The tests compared processed and unprocessed speech of the same peak power. Listeners were subjected to a constant level of band-limited white noise (100 to 5000 hertz). Speech level was increased in 3 - decibel increments to give MRT scores ranging from very poor (about 60%) to excellent (nearly 100%).

The graph shows that processing, at 80% intelligibility, is equivalent to using 5.2 times the audio power (7.2 dB). At 90% it is equivalent to 7.4 times the power (8.7 dB).

The preamplifier is mechanically and electrically interchangeable with the AM-4153 preamplifier used in existing shipboard systems throughout the Fleet. It is also designed to be part of the entirely new announcing system being developed by Newport News Shipbuilding Company for CVNs under construction.

As a retrofit on existing ships the new unit offers these advantages: For the 5MC flight deck system, where intelligibility is sometimes inadequate on some parts of the deck even with six kilowatts of audio power, using the new preamplifier will bring a considerable improvement without adding more power amplifiers and loudspeakers. For the 1MC (general) and 3MC (hangar deck) systems, where intelligibility is usually good enough if the equipments are operating properly and not overloaded, the new units will allow most loudspeakers to be turned down one 6dB notch (1/4 power). The result will be slightly better intelligibility and a reduction of audio power by a factor of nearly four. Currently overloaded amplifiers will then be running conservatively, and it will be possible to shift the load from a failed amplifier to an operational one without overload, while the failure is being corrected.

As a component of the new CVN system the preamplifier may reduce the number of loudspeakers required in some spaces and will reduce the total audio power needed. The actual saving in power can be found by comparing alternative designs using a computer program written by Code 441. Two runs of an earlier version of this program, for CVN 71, showed a power saving of about 2 to 1 on the 1MC/3MC system.

Continued next page.....

Other possible applications for analog speech processing include various sound systems, voice radio links, and any voice communication system where intelligibility can be impaired by noise added to the channel or surrounding the listener. An interesting application now being investigated by Code 441 is a decrease in peak audio power on submarines to reduce the sound transmitted through the hull. BY CRAIG R. ALLEN, 10-4-84 ♦

THE GREATNESS OF A TEACHER

The greatness of a teacher is not measured by how much they know but by how much they share.

Rev. Jesse Jackson

Glynn Walden, Engineering Manager of KYW Radio in the Philadelphia area, was one of our team leaders during a class exercise. Note his involvement with his team. Glynn obviously loves to share what he knows. ♦

WHERE COMPUTERS ARE HEADED

"The goal is a dramatic one: to grow computer circuitry in biology labs from living bacteria, producing microprocessors with 10 million times the memory of today's most powerful machines.

"If the new technology is successful, future circuits will be designed from specially tailored groups of organic proteins the size of molecules that would serve as electronic memory and switching devices in chips.

"Some of the groundwork needed to turn molecules into computers has already been laid. Scientists at the University of Mississippi and at Warwick University in Britain are experimenting with organically grown chemicals that turn on and off like transistors and display other electrical properties like those of silicon."

Pipe dreams -- fantasy -- well, these quotes are from a perfectly serious story in the January 7th issue of "U.S. News and World Report," page 50. "Some researchers are convinced a working model of an organic computer could be completed by the mid 1990s." This reader already knows one called Heyser. ♦

MOTTO AT SYN-AUD-CON

If I cannot be free,
To do such work as pleases me,
Near woodland pools and under trees,
You'll get no work at all;

For I would rather live this life
and die a begger or a thief,
Than be a working slave
with no days free.

William H. Davies (1871-1940) ♦

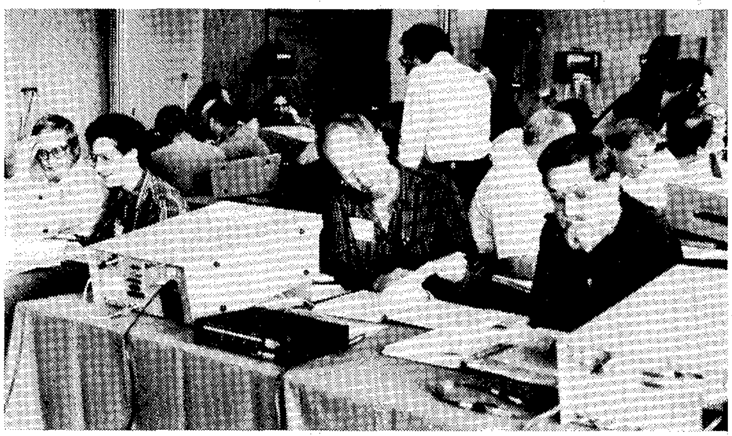
SMILE

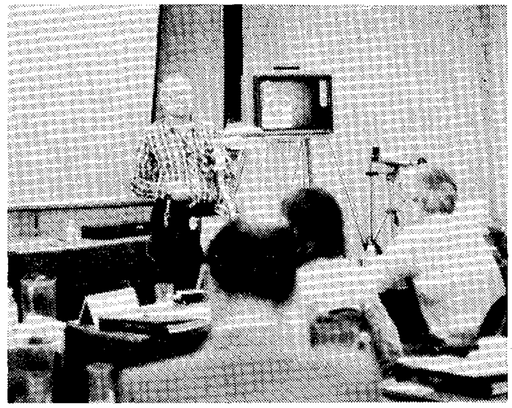
In a recent "AES Journal," "Exhibit Previews," a cable manufacturer stated:

Also shown will be a new generation of interconnecting cables that align the signal in amplitude and phase for improved sound quality. ♦

SPECIAL TEF® CLASS - ATLANTA - NOV. 1984

There are many steps in the TEF ladder. All of us, a few years back, were on step one of how to even get a measurement up on the screen. Many TEF users have now had a chance to discover the wondrous world of array and room measurements done for the first time with such accuracy and resolution. This means that future TEF workshops will need to be divided into two parts:





1. The first part for those interfacing TEF technology for the first time so they can learn the analyzer, the basic equations, and how to make each of the simpler measurements.

2. The second part will deal with the advanced concepts embodied in the Heyser and Peutz disks and the shared knowledge of the advanced participants.

Don Eger

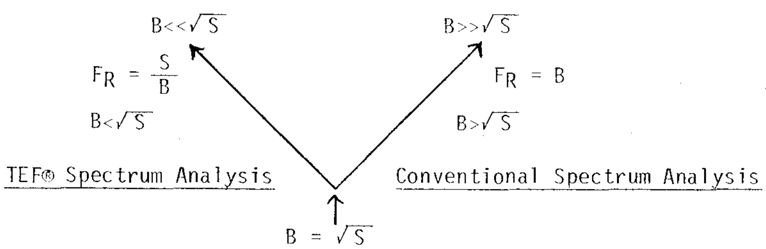
The Atlanta class was a good model for part one. We concentrated on TEF basics. Don Eger presented a very clear view of the fundamental difference between TEF measurements and conventional wave analyzer measurements.

When the bandwidth B of the tracking filter is equal to the square root of the sweep speed S

$$B = \sqrt{S}$$

you are then at the junction point between conventional wave analyzers and TEF analyzers (see Figure #1).

Fig. 1: Spectrum Analysis Frequency Resolution



Lowest frequency accurately measured is: $L.F. = F_R/2$

Fastest sweep speed for "non leakage" is: $S = \frac{F_1(F_2 - F_1)}{200}$

We are finding that a major stumbling block in the minds of engineers using analyzers is the difference between bandwidth B and frequency resolution F_R. Note in the case of conventional wave analyzers: $B = F_R$

Because this relationship was so often the first one encountered in one's measuring career the temptation is present to carry it over into FFT and TEF type analyzers. In the FFT case:

$F_R = \frac{\text{Total Bandwidth}}{\text{Number of Lines}}$ so that a 400 line FFT looking from 0 to 20,000 Hz *can have* an $F_R = \frac{20,000}{400} = 50\text{Hz}$.

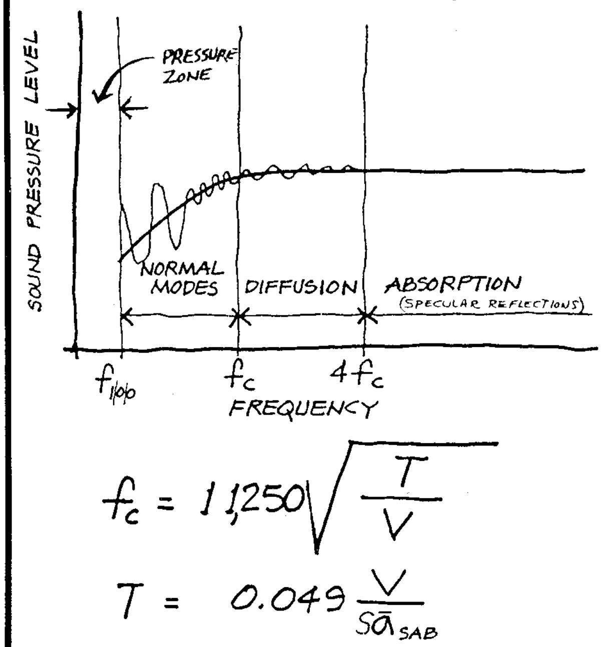
Continued next page.....

"Can have," that is, *if* the time bandwidth product is made equal to unity.

$$T_R \cdot F_R = 1.0 \qquad \text{Therefore:} \qquad T_R = \frac{1}{F_R} = 0.02 \text{ secs}$$

This means that your test signal must excite the device under test for at least 20 msec if this F_R is to be achieved. When impulse excitation is used, for example, the $F_R = \frac{1}{T_R}$ of the impulse.

Fig. 2: Controllers of Steady-State Room Acoustic Response



Courtesy of Bolt Beranek Newman, Inc.

In the TEF case: $F_R = \frac{S}{B}$ and $T_R = \frac{B}{S}$ Note the freedom gained to use any range of frequencies desired.

When ETC measurements are being made, it is desirable to set the sweep speed S so that:

$$S = \left(\frac{F_1\,(F_2 - F_1)}{200}\right)$$

Where: F_1 is the lowest frequency
F_2 is the highest frequency

For instance, if we are doing ETCs in a large hall and wish F_1 to F_2 to be 1000 to 1000 Hz, then:

$$S = \left(\frac{1000(2000 - 1000)}{200}\right) = 5000 \text{ Hz/sec}$$

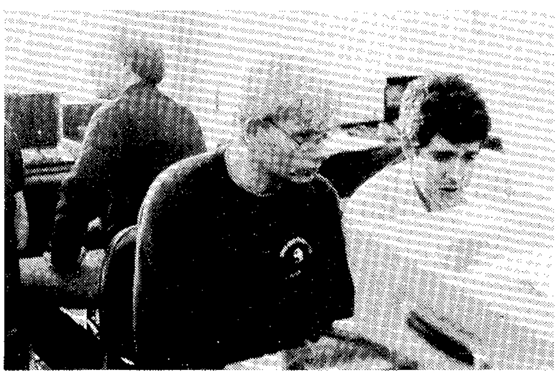
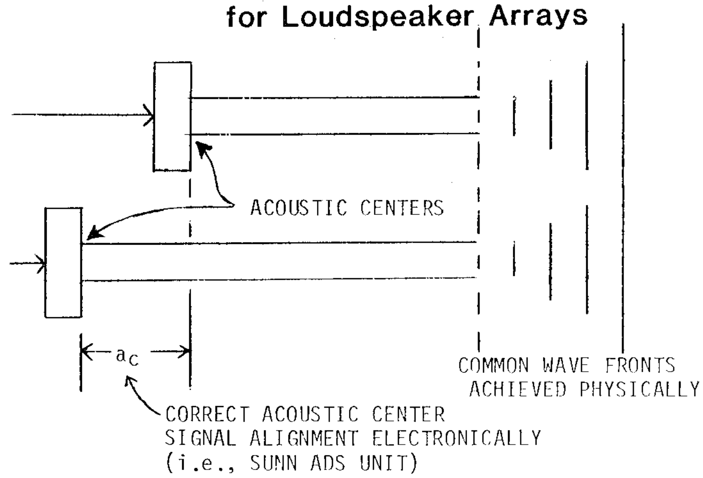


Fig. 3: Signal Alignment Practices for Loudspeaker Arrays



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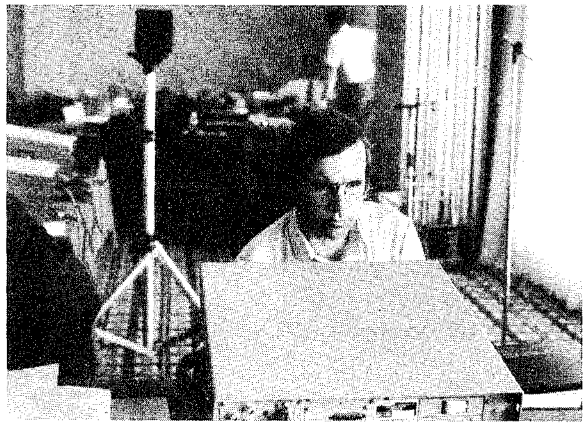
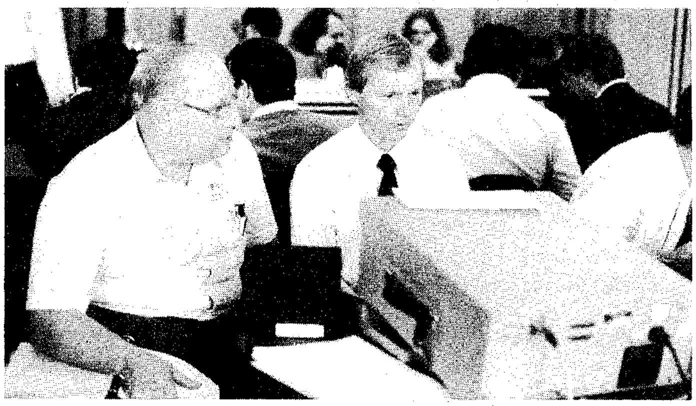
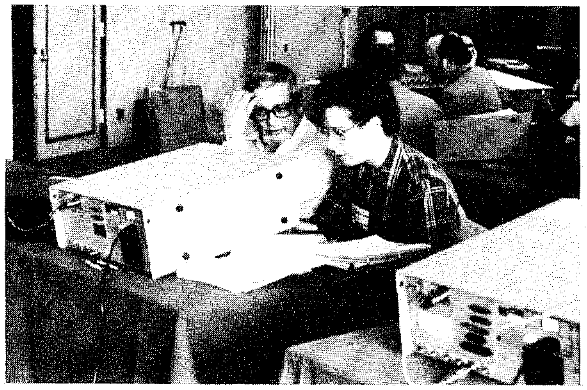
Don Keele

Don Keele presented valuable data with regard to sound fields in small rooms, pointing out that below F_{pp} (see Figure #2) the room acts as a pressure coupler and excellent bass response can again be expected, i.e., 1/10 of f_c. Don also pointed out what we believe will become a well-accepted practice when he suggested that the acoustic wavefronts of loudspeakers are what need to be aligned *physically* and that the acoustic centers should then be aligned electronically (see Figure #3).







This workshop was intense, basic, practical, and was topped off by a measurement session in the famous Fox Theater on Peachtree Street in Atlanta with Dr. Patronis. Dr. Patronis also presented the completed version of his "footprint" loudspeaker mapping program as well as a full discussion of absorption measurements.

When Deward Timothy of Salt Lake City was asked what he was doing with his TEF analyzer, his reply was, "Do you have three days?". He has found the TEF is magic on loudspeaker array adjustment and for spec writing. ♦

A MESSAGE TO.....NON-TEF® OWNERS

Do you skip over articles that show TEF measurements...or articles that describe how to make certain TEF measurements...thinking "this is not for me" because you don't own or plan to own a TEF analyzer in the near future?

MANUFACTURERS WHO OWN TEF ANALYZERS

Altec/Lansing	Electro Voice
Community Light & Sound, Inc.	J. W. Davis & Company
Crown International, Inc.	JBL Incorporated (budgeted this year)
Emilar Corporation	Shure Brothers Inc.

Reconsider please! The manufacturers listed here are testing and making measurements of their equipment. More and more of their literature, specifications, and ads will use TEF measurements. You need to know how to interpret them.

TEF is not for the technical elite alone. Just as the HP9100 (1968) made clear that the HP35 was coming (1972), so the current TEF analysis foretells a whole new generation of loudspeakers and microphones. It is only the beginning of a new way of thinking and measuring. ♦

NEW TEF® OWNERS

Stig Thodenius
Sveriges Rikstradio AB
S-105 10 Stockholm
Stockholm, Sweden

University of Southern CA
Cinema-TV
850 W. 34th St
Los Angeles, CA 90007

Bill Moseley
Custom Sound, Inc
6105 Yource Dr.
Shreveport, LA 91105

Arizona Public Service
Cholla Power Plant
Joseph City, AZ 86032

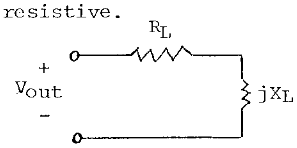
Tim Hill
Brigham Young University
685 E. University Parkway
Provo, UT 84602

Steve Orfield
Orfield Assoc.
4551 Bloomington Ave S
Minneapolis, MN 55407

A RIGOROUS DEFINITION OF OUTPUT POWER

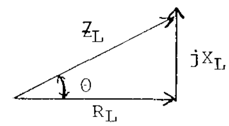
Comment by Don: Our sincere thanks to Craig Allen of NOSC for correcting my mental lapses with elegant clarity. Note that *Sound System Engineering*, page 23, Fig. 2-6, has Craig's correct equation (P.F. = cosΘ). I guess it shows me I need to re-read the yellow book occasionally.

While looking up your address, I ran across your Tech Topic Vol. 11, No. 10. Since you are a stickler for technical accuracy, a sworn enemy of the "voltage decibel," etc., I thought you might appreciate knowing about some errors in this Tech Topic. They arise from an attempt to be general and consider complex impedances instead of the usual assumption in audio work that everything is purely resistive.



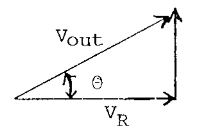
Under Power in the Load you say $P = \frac{V_{out}^2}{R_L}$ where $R_L = |Z_L| \cos\Theta$.

The latter equation implies that you are using the series equivalent circuit for the complex load impedance Z_L. But $P = V^2/R$ only when V is the voltage *across the resistor alone*. The terminals of the series equivalent resistor are not accessible. The only measurable voltage is that across the entire Z_L.



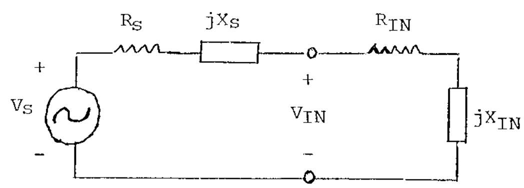
Using that V_{out}, P is not $\frac{V_{out}^2}{R_L}$.

The voltage phasor diagram shows that the voltage across the series equivalent resistor is $V_R = V_{out} \cos\Theta$.



Thus $P = \frac{V_R}{R_L} = \frac{(V_{out} \cos\Theta)^2}{R_L}$. Using $\cos\Theta = \frac{R_L}{|Z_L|^2}$ a little algebra gives another useful equation: $P = \frac{V_{out}^2}{|Z_L|} \cos\Theta$.

Depending on the kind of impedance meter you use, it may be easier to measure impedance magnitude $|Z_L|$ and power factor ($\cos\Theta$) instead of series equivalent resistance and reactance.



A similar error appears under Measuring AIP. The voltage divider equation shows that

$$V_S = V_{IN} \frac{Z_S + Z_{IN}}{Z_{IN}}$$

Where Z_S and Z_{IN} are complex numbers, not just magnitudes.

This is not the same as $V_{IN} \frac{R_S + R_{IN}}{R_{IN}}$ unless *both* X_S and X_{IN} are zero. Actually the way to find V_S is simply to measure the open-circuit voltage of the source device without load, using a high-impedance voltmeter. Then AIP $= \frac{V_S^2}{4R_S}$. It is worth noting that the available power from a device having a complex internal impedance is the power delivered to a *conjugate*-matched load. That is, $Z_L = Z_S^* = R_S - jX_S$. The circuit is then in series resonance, the net total reactance is zero, and the power delivered to the load is $\frac{V_S^2}{4R_S}$. That is not a realistic condition in audio work, perhaps, but it *is* the definition of available power. ♦

WHY MEASURE Z INSTEAD OF R

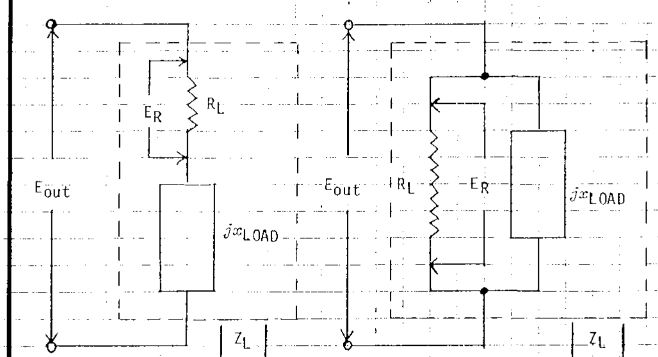
In looking at the loudspeaker as a load for a power amplifier, it can be modelled as either a pure resistance in series with a reactive component or as a pure resistance in parallel with a reactive component.

Often we cannot access separately the resistive and reactive parts but we can measure the total Z_L that they represent, even when we don't know the exact circuit configuration. (See the illustration at right.)

Note that the equation as derived in the illustration results in the true power dissipated in the load and allows a comparison between it and W = EI so that the reactive power (wattless power) can be computed.

It is in measurements of this type that the ability of the TEF® analyzer to provide us with the complex impedance plot is so useful. As most of us are rediscovering, loudspeakers are indeed complex impedances and many amplifiers were not designed to look at such loads. ♦

A MORE RIGOROUS DEFINITION OF OUTPUT POWER



$$W = \frac{(E_R)^2}{R_L} = \frac{(E_{out} \cos\Theta)^2}{R_L} \qquad \cos\Theta = \frac{R_L}{|Z_L|}$$

$$W = \frac{\frac{(E_{out})^2 R_L \cos\Theta}{Z_{LOAD}}}{R_L} = \frac{(E_{out})^2}{|Z_{LOAD}|}(\cos\Theta)$$

CROWN MICRO-TECH 1000

For those of us who have to carry amplifiers around the country with us, reliability, small size, and easy access to service or replacement stand high on our list of desirable specifications. The fact that we can have *all of that* and Crown quality is as near Utopia as you're likely to get. The following release from Crown indicates they're still way out in front of whatever is in second place.

CROWN INTRODUCES MICRO-TECH 1000 STEREO POWER AMPLIFIER

The Crown Micro-Tech 1000 is a miniaturized high-technology stereo power amplifier for professional sound reinforcement and studio monitoring. This amplifier provides very high power within a low-profile package.

The Micro-Tech can deliver 1000 watts *continuous average power* (Editor's italics) in mono mode at less than 1 percent THD, into 1 or 4 ohms. A "parallel mono" switch combines the outputs of both channels to make a monophonic amplifier capable of 1000 watts into 1 ohm. By adding an internal jumper for the "bridge mono" configuration, the user can obtain 1000 watts into 4 ohms. In stereo mode, the Micro-Tech provides 250 watts per channel into 8 ohms, or 350 watts per channel into 4 ohms.

The Micro-Tech 1000 uses an "Output Device Emulator Protection" (ODEP) circuit which simulates the output transistors. With this circuit, the amplifier can detect and compensate for overheating and overload. The unit is also protected against output shorts, open circuits, mismatched loads, overall overheating, and high-frequency overloads.

Efficient heat sinking and a self-contained forced-air cooling system prevent overheating and prolong component life. The direction of airflow can be reversed, if necessary, to work with the rack cooling system -- a feature unique to Crown. The dust filter located on the front of the unit is easily removed for cleaning or replacement.

Inputs are balanced 1/4" phone jacks with adjustable gain. Outputs are 5-way banana jacks for minimum power loss.

Hum and noise are 105 dB below rated output (A-weighted). Harmonic distortion is less than 0.05% from 20 Hz to 1 kHz and increasing linearly to 0.1% at 20 kHz delivering 250 watts into 8 ohms, per channel. I.M. distortion is less than 0.05%, and slewing rate is greater than 13 volts per microsecond. ♦

PHASE COHERENT? LOUDSPEAKER

During our fall tour we had occasion to measure a loudspeaker we were told was "absolutely *time* aligned and phase coherent." Naturally, we are interested in all such devices. We'd like to pass on to you the TEF® measurements we made without further comment and, mercifully, without product identification. As Bernoulli said of Newton upon viewing the solution to the BRACHISTO-CHRONE (a cycloid), which Newton sent anonymously, "Ah! I recognize the Lion by his paw." Well, here are the knock-kneed, pigeon-toed paws for this loudspeaker. Please note that everything on the ETC display of Figure #1 *belongs to the loudspeaker*. No room reflections are included. The second ETC (Figure #2) is with the "tweeters" cut off.

Vertical: 6dB/div with base of display at 30.1dB
0dB is located at .00002 Pascals

Horizontal: 4730 microseconds or 5.3449 Feet to
11023 microseconds or 12.4565 Feet
scale: 1.9444E+00 Feet/inch or 7.6550E-01 Feet/cm.
1720 microseconds/inch or 677 microseconds/cm.

Line Spacing: 15.773 microseconds or 1.78235E-2 Feet
Line Width: 21.4513 microseconds or .02424 Feet

Sweep rate: 5009.55Hz/Sec

Sweep range: -31699.70Hz to 31699.70Hz

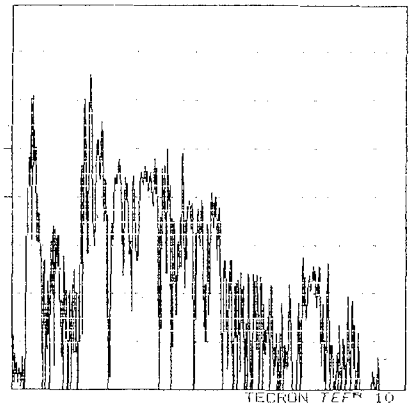


FIG. 1: *The entire display belongs to the loudspeaker.*

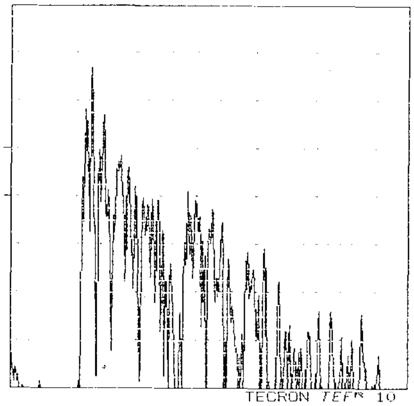


FIG. 2: Same speaker but with tweeters cut off.

The Nyquist (Figure #3) indicates that, in this case, alignment is a relative term. A smooth circle would be perfect (see the TOA M8230 for a Nyquist with few problems).

Finally, the end result is the response curve in Figure #4. Our conclusion: perhaps the designer is not a new audio Messiah but a mesmerist. When typical users see what"s actually going on, they react like the famous smoker who was asked if he had read the Surgeon General's report on smoking. Yes, he had. Was he going to quit smoking? No! He was going to quit reading. These users sometimes feel they are going to quit measuring. Incidentally, speech on this system sounds as bad as it measures.

Resolution: 1.1279E+00 Feet & 1.0019E+03Hz

6dB of automatic screen gain.

Frequency range: Auto 0.00Hz to 19998.10Hz

Time of test: 5437 microseconds, 6.2004E+00 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 5.0000E+00Hz

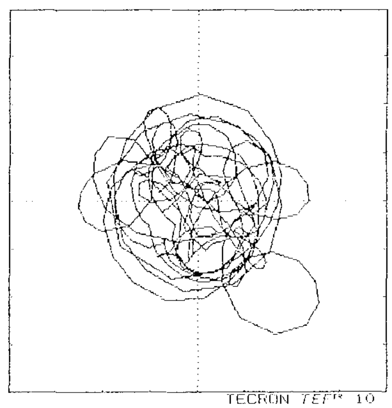


FIG. 3: Niquist display. A smooth circle would indicate perfect alignment and phase coherent.

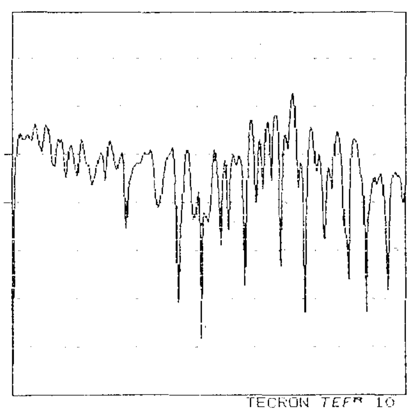


FIG. #4.

Vertical: 6dB/div with base of display at 42.1dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 19998.10Hz
scale: 5467.68Hz/inch or 2152.63Hz/cm.

Resolution: 6.7671E+00 Feet & 1.6699E+02Hz

Time of test: 5487 microseconds, 6.2004E+00 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 3.0000E+01Hz

In all fairness, many contemporary music listeners *feel* it sounds good. At least, it's loud, and Dr. Patronis showed us in the Fox Theatre in Atlanta, with a little demo he performed, that one can't make valid judgments of quality at extremely high levels. But, with certainty, you can say it's loud! ♦

WORKSHOPS - SPRING 1985

TEF® - Basic Measurements

DON DAVIS • DON EGER • FARREL BECKER

March 26 - 27 Fee: $400

TEF® - Advanced Measurements

DON DAVIS • DON EGER
RUSS BERGER • JIM BROWN • FARREL BECKER

March 28 - 29 Fee: $400

HACIENDA HOTEL - LAS VEGAS, NEVADA
Fee for 4 days: $700

Grounding & Shielding

RODEWAY INN - BLOOMINGTON, MINNESOTA
ED LETHERT • TOPPER SOWDEN

May 14 - 16 Fee: $500

LEDE™ Control Room Design

SOUNDS INTERCHANGE - TORONTO, CANADA
CHIPS DAVIS • NEIL MUNCY • DON DAVIS

June 21 - 23 (Dates could shift)

Fee: $400 (U.S. Funds) 2 Days
$100 Optional 3rd day

To be scheduled:

- LOUDSPEAKER ARRAY WORKSHOP
- INSTALLING & TROUBLE-SHOOTING THE SOUND SYSTEM
- AUDIO FOR BROADCASTING

SEMINARS - SPRING 1985

Dallas, Texas

THE SUMMIT HOTEL
April 24 - 25

Kansas City, Missouri

INN @ EXECUTIVE PARK
May 8 - 9

Indianapolis, Indiana

HOLIDAY INN - I-70 East
May 21 - 22

Boston, Massachusetts

June 5 - 6

Toronto, CANADA

CONSTELLATION HOTEL
June 12 - 13
"On Location - June 14

RANDY KIMMEL

We don't have very many 16 year olds in our classes -- nor do we have very many at any age as sharp as 16 year old Randy Kimmel.

Randy works after school, weekends and during the summer for Russ O'Toole at Audio Electronics in Romeoville, Illinois (a suburb of Chicago). When Russ called to register Randy, he asked us to keep an eye on Randy and help him along if he needed it. Randy was "with it" all the time. ♦

THE CHURCH BUILDING (A Sound Contractor's View)

An edifice conceived by the congregation for the improvement of the worship service, designed by God's next of kin, the architect, that significantly exceeds the budget and when finally built, exhibits questionable esthetic beauty peculiar to the denomination and possesses unacceptable acoustic properties which were forseen by the architect who corrected the problem by approving a sound reinforcing system at the lowest possible bid because the project was already over budget due to the costs of the questionable esthetic beauty and now shall be corrected by the congregation sound expert who was not originally consulted. But the sermons still cannot be understood because of some unknown reason since everything possible has been tried. The only thing left is to pray for deliverance through God's next of kin, the architect who, hopefully may suddenly realize there are other kinfold of God's that specialize in the mysterious behavior of sound and, if contracted with, can deliver the congregation and allow them to understand the word God has been trying to convey for nearly 2000 years.

By Bob Fellers, Communications Company, San Diego, California ♦

GROUNDING & SHIELDING WORKSHOP

Staff: Ed Lethert & Topper Sowden

May 14-16, 1985
Minneapolis, Minnesota

The most asked for special subject in Syn-Aud-Con classes is grounding and shielding of circuits. Before one tackles this seemingly simple basic subject, they'd be wise to first discover the voluminous literature written by some of the brightest engineering minds in industry.

We are very fortunate to have instructors conversant with this literature, experienced in its real life application, and expert enough to extend its application into the digital age. We already know that the grounding techniques used in recording studios today cannot be safely used in a fully digital studio.

Ed Lethert sent us the preliminary *partial* outline below indicating some of the subjects to be covered in this 2-1/2 day workshop.

GROUNDING & SHIELDING

THE INTERFERENCE PROBLEM

- What is ElectroMagnetic Interference
- What is ElectroMagnetic Compatibility
- What is Normal Mode and Common Mode Noise

GROUNDING (to Earth)

- The Power Ground and the Signal Ground
- What is it? How important is it? How is it achieved?

ZERO VOLTAGE REFERENCE

- The Most Important Issue!
- The Signal Zero Reference Point for low frequency (typically analog) systems
- The Signal Zero Reference Plane for high frequency (typically analog) systems
- Connecting the Zero Reference to Ground (earth)
- Coping with Multiple Zero References for Signal and Power Circuits
- (This will address systems as a whole, not individual components such as consoles)

CROSSTALK IN CABLES

- Intruding signals and Emitted Signals
- Why cables are shielded
- Why cable wire pairs are twisted
- Coaxial Cable vs. Twisted Pairs
- Why the microphone cable can't be in the same raceway with the 70 volt loudspeaker line
- How to calculate the potential interference between any two cable circuits
- Fiberoptic links and the Zero Reference

SHIELDING OF ROOMS

- When and How - An Overview

ELECTROSTATIC DISCHARGE

- How to deal with it in the operational environment
- How to deal with it in the maintenance environment

Ed Lethert, Director of Systems Engineering
Michaud Cooley Erickson & Associates, Inc. ♦

NEW ADVENTURES

Chris Foreman - Chris was with Stanal Sound when we first met him years ago. He mentioned at that time that he would like to write. He has arrived. Chris' new job is Editor of "Sound & Communications," whose new owner is publisher of "Pro Sound News."

Richard Lee - Rich is now with Sony Professional Audio Division as National Product & Systems Manager. He previously worked at Criteria as Vice President/General Manager, Compass Point, and as a consultant for LEDE™-type control rooms. ♦

Carl Dorwalt - Audio veteran, Carl Dorwalt is now Vice President of Marketing at Bogen. ♦

"ON LOCATION" & ORLANDO CLASS

The "On Location" measurement workshop at Bob Carr Auditorium had the opportunity to attend the full-dress rehearsal of Verdi's Rigoletto with a good professional cast; measure in the auditorium with the TEF® analyzers on Friday; and, finally, to move about during the evening of the full performance and compare seats previously measured.

SYNERGETIC AUDIO CONCEPTS

"ON LOCATION" MEASUREMENTS
November 9, 1984
BOB CARR AUDITORIUM

Willard P. Armstrong
7530 Twelve Oaks Blvd.
Tampa, FL 33614

William L. Bencsik
Bencsik Associates
3730 N.E. 42nd Street
Ocala, FL 32670

Steve Bushelman, Jr.
National Electronic Systems, Inc.
2004 Ft. King Road
Dade City, FL 33525

Joseph C. Cook
(U. S. Government)
303 Knoll Street N.W.
Vienna, VA 22180

Joel Fears
(NASA)
722 Mercedes Avenue
Daytona Beach, FL 32014

Bill Garner
House of High Fidelity
599 Fifth Avenue South
Naples, FL 33940

Hellmuth Kolbe
Ing. buro f. akustik u. schallshutz
beratendering AES
Zielackerstrasse 6
CH 8304 Wallisellen, SWITZERLAND

Michael Lowhorn
Stereo Sales, Inc.
P. O. Box 6297
Tallahassee, FL 32301

Richard Menasco
Stereo Sales, Inc.
P. O. Box 6297
Tallahassee, FL 32301

Charles S. Meyer
Pyrotronics
5168 Douglas Avenue
Altamonte Springs, FL 32714

Lawrence E. Meyer
(Bencsik Associates)
9362 Southampton Place
Boca Raton, FL 33434

Gerald Phillips
Audio Video Analyst
944 Volusia Avenue
Daytona Beach, FL 32014

Robb Resler
Walt Disney World
Entertainment Division
P. O. Box 40
Lake Buena Vista, FL 32830

Robert Sexton
Audio Video Analyst
944 Volusia Avenue
Daytona Beach, FL 32014

Stig Thodenius
The Swedish Radio Company
A1/21
Stockholm, SWEDEN S-105 10

Donald J. Washburn
The Audio Bug, Inc.
13140 N.W. 7th Avenue
North Miami, FL 33168

Carolyn Davis
Synergetic Audio Concepts
P. O. Box 669
San Juan Capistrano, CA 92693
New Phone # (714)728-0245

As a result, we heard an excellent operatic performance, some spectacular spatial effects with shifting aural images, and their frequency dependence.

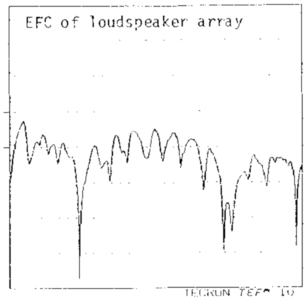
Vertical: 12dB/div with base of display at 30.1dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 10001.20Hz
scale: 2734.43Hz/inch or 1076.55Hz/cm.

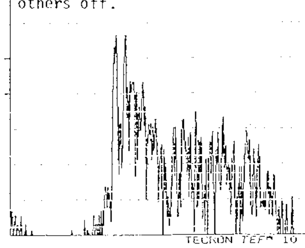
Resolution: 6.7671E+00 Feet & 1.6679E+02Hz

Time of test: 69600 microseconds, 7.8648E+01 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 3.0000E+01Hz



EFC of loudspeaker array



The loudspeaker array. 2 horns aimed at middle seating. All others off.

Vertical: 6dB/div with base of display at 42.1dB
0dB is located at .00002 Pascals

Horizontal: 60000 microseconds or 67.8 Feet to
86601 microseconds or 97.8588 Feet
scale: 8.2184E+00 Feet/inch or 3.2356E+00 Feet/cm.
7272 microseconds/inch or 2863 microseconds/cm.

Line Spacing: 66.6683 microseconds or 7.53352E-2 Feet
Line Width: 90.6689 microseconds or .102456 Feet

Sweep rate: 2030.90Hz/Sec

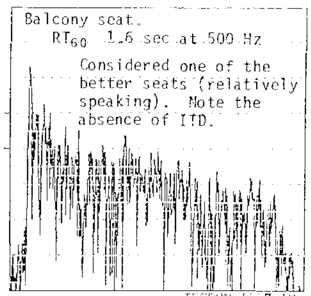
Sweep range: 0.00Hz to 14999.60Hz

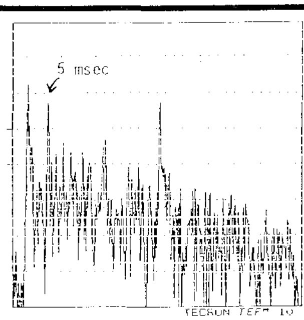
Hellmuth Kolbe, who flew over from Switzerland early to be on hand for the Atlanta TEF class, helped in making the evaluation of Orlando's Carr Auditorium. For instance, when Hellmuth arrived Friday a.m., we commented that no bass got to the audience area. He said, "Of course not!", and pounded his fist on the side walls of the auditorium. The walls were perfect bass absorbers.

Continued next page.....

One factor always stands out to us as we do these "live music" workshops and that is that "live music," even in a bad hall (not to say the Bob Carr was a bad hall) is so superior to the best recorded efforts and that poor performances are poor performances in either format. While we love our records, we increasingly try to

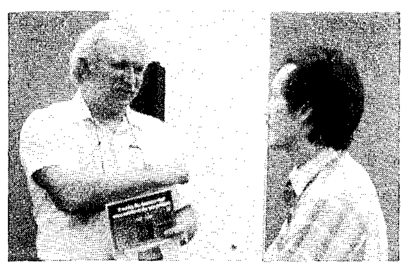
```
Vertical:   6dB/div with base of display at    42.1dB
            0dB is located at  .00002 Pascals

Horizontal:   100000 microseconds or  113 Feet to
              277186 microseconds or  313.22 Feet
     scale: 5.4743E+01 Feet/inch or  2.1552E+01 Feet/cm.
            48444 microseconds/inch or  19072 microseconds/cm.

Line Spacing: 444.075 microseconds or  .501805 Feet
Line Width:   603.942 microseconds or  .682455 Feet

Sweep rate:  1001.91Hz/Sec

Sweep range:    249.49Hz to   2501.36Hz
```






```
Vertical:   6dB/div with base of display at    42.1dB
            0dB is located at  .00002 Pascals

Horizontal:   100000 microseconds or  113 Feet to
              184019 microseconds or  207.941 Feet
     scale: 2.5958E+01 Feet/inch or  1.0220E+01 Feet/cm.
            22971 microseconds/inch or  9043 microseconds/cm.

Line Spacing: 210.573 microseconds or  .237947 Feet
Line Width:   286.379 microseconds or  .323608 Feet

Sweep rate:  1001.91Hz/Sec

Sweep range:    249.49Hz to   4998.45Hz
```

seek out the real reference source -- a live performance -- in order to properly calibrate that most marvelous analyzer of all, the human mind.



We are grateful to the people who make these valuable "On Location" measurement sessions possible by their attendance. They are great learning experiences. The sessions are an amazing amount of work for us to arrange but well worth it. We have held two now -- The Boettcher Performing Arts Center and Bob Carr Auditorium. In each case, we heard a memorable music performance and vastly increased our understanding of the concert hall. ♦

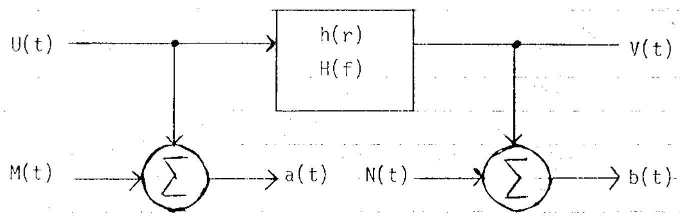
FFTs ARE NOT FOR ACOUSTIC MEASUREMENTS

The current consumer magazines, AES conventions, and various other points of sale are being inundated with marvelously annotated displays of erroneous mismeasured data taken with FFTs. The following explanation and objections apply not only to the "toy" category that you plug into your home computer but to the most expensive multichannel analyzers as well. Syn-Aud-Con is not a "tested in the home" organization but these abuses are becoming so widespread and some of the perpetrators are so belligerently ignorant that we feel compelled to warn our "grads" that anyone telling you that you can do meaningful acoustic measurements with an FFT should be sufficient cause for termination of the conversation.

FIG. 1:
Measuring Frequency Response With FFTs

$$H_1 = \frac{\text{cross spectrum from input to output}}{\text{input auto spectrum}}$$

$$H_2 = \frac{\text{output auto spectrum}}{\text{cross spectrum from output to input}}$$



$$H = Guv/Guu \quad \text{or} \quad Gvv/Gvu$$

$$H_1 = \frac{H}{1+(Gmm/Guu)} \quad \text{(underestimates response)}$$

$$H_2 = H\left(1+\frac{Gnn}{Gvu}\right) \quad \text{(overestimates response)}$$

$$H_1/H_2 = \text{Coherence Function}$$

The Fourier transform is only valid for *linear* systems.

The Fourier transform is only valid for *time invariant* systems.

FIG. 2: Basic Digital FFT Equations

I. DYNAMIC RANGE IN DECIBELS

dB = 20 Log(2 Exp N)

N = (Ln(10 Exp(dB/20)))/Ln 2

II. % FULL SCALE

%F.S. = 100(0.5 Exp N) = (lowest voltage/F.S. voltage)100

N = (Ln(%F.S./100))/Ln 0.5

III. FREQUENCY SPAN

H.F. ≤ (S.R.)/2

S.R. ≥ 2(H.F.)

S.R. is "sampling rate"

IV. FREQUENCY AND TIME RESOLUTIONS

$F_R \cdot T_R = 1.0$ = Time-bandwidth Product

$F_R = 1/T_R$; $T_R = 1/F_R$; $\frac{1}{T_{impulse}}$ = B.W.

F_R (for lowest freq.) = $F_R/2$

$T_{impulse}$ determines S/N

Brüel and Kjaer are always in the forefront of providing correct accurate data on instrumentation. The first illustration shows the influence of input and output noise on FFT measurements. The second illustration provides definitions for basic digital terms.

FFT Analysis and the Time-Bandwith Product

Apparently, a much misunderstood parameter relevant to all forms of analysis is the *time-bandwidth product*. The time-bandwidth product is the simple mathematical statement that $F_R \cdot T_R = 1.0$.

Where: F_R is the frequency resolution in Hz
T_R is the time resolution in secs

Thus: $F_R = \frac{1}{T_R}$ and $T_R = \frac{1}{F_R}$

FFTs and the Time-Bandwith Product

The Gen-Rad 2512 FFT analyzer handles this problem in the following manner. They specify an "input frame length" and they define it as:

Input Frame Length = 1024*/2.56 x freq.

*The 2512 is a 10 bit analyzer and $2^{(10)} = 1024$.

They then further state that "input frame length is synonymous with the term 'time window'."

If, for example, we decide on a total analysis bandwidth of 20,000 Hz, then their equation provides a time window of:

$$T_W = 1024/2.56 \times 20{,}000 = 0.02 \text{ secs} \quad \text{(20 msec)}$$

Referring to our fundamental relationship of: $F_R = \frac{1}{T_R} = \frac{1}{0.02} = 50$ Hz

We find that what the equation has done is to make sure that the T_R is correct for a 20,000 Hz bandwidth divided by the 400 lines of resolution in their analyzer.

$$\left(\frac{1024}{2.56}\right) = 400$$

Continued next page....

The Problem Arises

Here's how the confusion begins. Suppose one sets up the FFT for this bandwidth (20,000 Hz) and this input frame length (20 msec) but instead of using a steady state signal such as white or pink noise, we were to choose an impulse signal. Now we would want this impulse signal to equally excite all frequencies up to 20,000 Hz so it would *have to be*

$$\frac{1}{20{,}000\ \text{Hz}} = 0.00005\ \text{secs} \qquad (50\ \text{usecs})$$

Ah! But here's the "Catch 22." The analyzer now sees useful signal for only 0.00005 secs and listens to the noise present in the environment for .02 secs.

How to Optimize F_R and T_R

TEF® analysis fortunately has the answer to this dilemma. In TEF, $T_R = \frac{B}{S}$ and $F_R = \frac{S}{B}$

Where: B is the actual tracking filter bandwidth

S is the sweep rate in Hz/sec for both the swept oscillator and the tracking filter.

Several remarkable "energy" relationships fall out of these simple parameters. If we wish a 20,000 Hz bandwidth, we merely call for a sweep over that range of frequencies. Suppose we choose a sweep rate of S = 5000 Hz/sec. Let's further suppose that we want a time window of 4 msec (0.004 sec). Then our F_R becomes:

$$F_R = \frac{1}{0.004} = 250\ \text{Hz}$$

If we choose a T_R of 0.02 sec, we could then have an F_R of $\frac{1}{0.02} = 50$ Hz.

Note that with TEF we are free to *pick our frequency range* totally *indepdendent* of our F_R and T_R, whereas with the FFT impulse technique the impulse itself becomes the controlling factor.

Constraints on All Measurements

In *practical acoustic measurements* in the field, what we seek is freedom from nearby reflecting surfaces while we examine the direct sound level response, L_D, of our arrays. In order to do this, we need to establish a time window or time resolution, T_R, that is shorter than the travel time for the interfering reflection we are trying to avoid. This is quickly established by measuring on the ETC display the time interval between L_D and L_R and using that information as the basis for our T_R. Our T_R instantly determines our F_R ($F_R = 1/T_R$) and we proceed to set up our EFC parameters of S and B accordingly.

Both Brüel and Kjaer and Tecron have pointed out that the demarcation between TEF and normal swept spectrum analysis, in addition to the ability to offset the tracking filter from the oscillator in relative time (i.e., delay the tracking filter's sweep), is the relationship: $B = \sqrt{S}$.

When the actual filter bandwidth (to differentiate from the *effective* filter bandwidth) is greater than the square root of the sweep rate, then you are engaged in normal spectrum analysis. When the actual filter bandwidth is less than the square root of the sweep rate, then you have TEF analysis conditions. Note that in TEF analysis

$$F_R = \frac{S}{B}$$

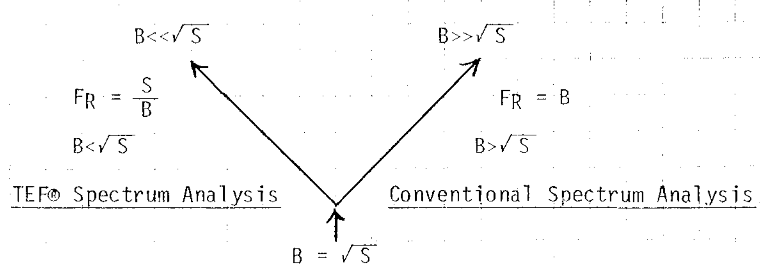
whereas in normal spectrum analysis

$$F_R = B \qquad \text{(See Fig. \#3)}$$

A universal measurement constraint in all acoustic measurements is the 1/F factor. If we wish to accurately measure down to some low frequency F, then we must have a T_R that is at least as long as 1/F. This constraint applies to anechoic chambers, TEF, or any other analysis form you may happen to choose. For example, if you wish to measure down to 30 Hz, then

$$T_R = \frac{1}{30} = 33.3\ \text{msec}$$

FIG. 3: Spectrum Analysis Frequency Resolution



Lowest frequency accurately measured is: L.F. = $F_R/2$

Fastest sweep speed for "non leakage" is: $S = \frac{F_1(F_2 - F_1)}{200}$

Continued next page...

This also translates into not having a path length for any reflection shorter than

$\frac{c}{F}$ Where: c is the velocity of sound in ft. or m per sec. (i.e., $\frac{1130}{30} = 37.7$ ft.)

The TEF user merely locates a large space. The standard spectrum analysis requires a large anechoic chamber and we'll leave it to your imagination how much effective absorption is present at 30 Hz, even in a large chamber. (The TEF user has 60 dB of discrimination against the unwanted reflection outside of his time window.)

Signal-to-Noise Ratios (S/N)

Let's revert one more time to our FFT illustration where we use a 50 usec impulse in order to obtain a 20,000 Hz bandwidth. The FFT will be set to an input frame length of 20 msecs in order for the display on the screen to be in accord with the frequency span chosen. The TEF can easily be set to sweep the same range in one, two, three or more seconds. The FFT, when triggered by the impulse, must search over the entire time interval of the input frame length looking for the 50 usecs of actual signal. The TEF analyzer has the tracking filter "locked" to the desired signal throughout the entire sweep time, i.e., it has preknowledge of where to look and does so. This results in a signal-to-noise advantage for the TEF of

$$S/N = 20 \, \text{Log}\left(\frac{1 \text{ sec}}{0.00005 \text{ secs}}\right) = 86 \text{ dB}$$ over the FFT impulse test.

Conclusion

FFT analysis, while useful in some electronic applications, *is not* the proper tool for acoustic measurements (it can be used to measure ambient noise levels) for the following reasons:

1. The FFT measurement is invalid if the system is not linear.
2. The FFT measurement is invalid if the system is not time invariant, i.e., not affected by temperature, pressure, or other parameters.
3. FFT measurement cannot provide short T_W and detailed F_R at the same time.
4. FFT measurements, where short T_Rs are required, suffer major S/N disadvantages.

The Fourier transform, a degenerate form of the Heyser transform, is without doubt useful and widely utilized. Syn-Aud-Con is in no way trying to deprecate it, and when acoustic "one port" measurements are required, it is a tool readily chosen by us. As indicated at the beginning of this article, it is the misapplications and misrepresentations of its usefulness in sound system analysis that we protest. Your money is too hard to come by to spend on "toys" rather than tools. ♦

FFT ANALYZERS

We note with interest that the October 1984 AES Convention features a session on the use of FFT analysis in sound reinforcement work. We were immediately reminded of the wise man who stated:

A cup of wine in a barrel of sewage gives you sewage.
A cup of sewage in a barrel of wine gives you sewage.

This has been "transformed" by computers to: "Garbage in, garbage out"

With these profound thoughts, we leave the review of this momentous session to the contemporary consumer magazines who no doubt will now supply their equipment reviewers with this newest Hi Tech tool for the mis-examination of advertisers' products.

WHAT IS FREQUENCY?

I. The *IEEE Dictionary* definition: *The number of periods per unit time.*

II. For a time interval of 1 sec: 1 period = 1 cycle = 1 Hz = 2π radians = 360°

III. Thus, frequency is:

- Periods per second
- Cycles per second
- Hz
- Radians per second
- Degrees per second ♦

REMARKABLE RECORD FROM SWEDEN

Lennert Nilsson of the Swedish Radio Company attended our Nashville LEDE™ Workshop at Acorn Studios. He brought along a tape that thrilled the class and many asked how they could obtain a copy.

Recently, we received a letter from Lennert along with information about the Opus 3 recordings and six *outstanding* recordings. Syn-Aud-Con recommends without reservations of any kind that you write for their catalog and make your choice. The recordings are:

1. Technically and musically exceptional.

2. The quietest record surfaces we have ever encountered.

and 3. Their spatial geometry is so exact on a good reproducer (we are using the new TEF® designed Spica system) that images take on multi-dimensions in the space between loudspeakers.

If these recordings don't jump out into the room dimensionally as well as tonally, then there's something wrong with your music system.

THE SWEDISH RADIO COMPANY
RADIODIFFUSION SUEDOISE
SCHWEDISCHER RUNDFUNK

Date *30th October, 1984*
Your date

SVERIGES RIKSRADIO AB

Dear Don and Carolyn,

Regarding your great apreasiation of the "Segovia"-type of music I brought with me to the LEDE-workshop in Nashville, I promised to send you a copy. I have talked to Jan-Erik Persson who is responsible for the recording and he is pleased to give you the included records.

Many of the partisipans showed interest to get hold of a similar mastercopy as the one they heard. The mastercopy is aveable on direct request from Jan-Erik if 100 $ is prepaid.

Enclosed you will find information about the recordings and Jan-Erik's address.

Yours sincerely,

SVERIGES RIKSRADIO AB

Lennart Nilsson

INFORMATION ABOUT THE RECORDING

The choir-piece was made in a church built in the mid-19th century modelled on a medieval cathedral. The choir was positioned asymetrical on a big carpet in front of the alter. The microphone was placed some 8 meters symetrically in front of the choir. Half way to the microphone the carpet ends and the marble floor is exposed.

Microphone is Neuman SM-69. Tape recorder Modified Revox G-36 HS.

The guitar-quartet-piece was recorded in an old stone church from the 12th century. The guitarists were positioned according to the pictures on the cover of 7810. The distance to the musicians is 1.5 - 2 meters. The distance to the side walls is approx. 8-10 meters, to the rear wall (the alter) about 5-6 meters. The height above the players is approx. 20 meters. The floor is made of stone.

The microphone was AKG C-24 (tube-type), microphone preamps with tubes and Telefunken M-28 tape recorder.

Both recordings were made with Blumlein-technique, i.e., crossed figure of eights. This technique is used on all OPUS-3 records.

Firma Ljudinspelning/OPUS 3
Jan-Eric Persson
Kalendervägen 4
S-175 40 JÄRFÄLLA
Sweden
Tel. 0758-186, Pg 20 76 21-4

opus 3 ♦

PZM® CHALLENGE WINNERS

Richard Menasco of Stereo Sales in Tallahassee attended our Orlando class last fall.

We recognized his name from the PZM Challenge winners and asked Richard what his entry was that won. He said, "Which category?" It turned out that Richard and his brother, David, won an award in about every category.

On the same subject, Calf Audio in Ithaca, New York, has won the Grand Prize in the Dealer Category for three years straight. Last year Alfred Grunwell sent us a cassette of their winner. It is a winner! ♦

AES CONVENTION - FALL 1984

It became evident at the 1984 fall convention that the AES is no longer the chosen gathering place for those engineers interested in sound contracting. This year's convention witnessed a number of fascinating "tidal waves." The first of these is the desperate attempts at cramming FFT technology into field applications relative to sound reinforcement work. The results were, to be charitable, ludicrous. The unhappy manifestations are:

1. The many who are losing their money purchasing high promises and low results.

2. The young uninformed who are misled to spend wasted time up a blind alley.

Digital Concessions

Another "wave" is the dawning recognition on the part of serious workers in the digital recording field that they are in deeper trouble than they suspected and the time for confession is at hand. We do admire the honesty expressed by many of these workers. (See excerpts at the end of this article from the paper given by Dr. Lagadec and Dr. Stockham at the Paris AES Convention.)

Psychoacoustics

Surprising to us is the number of psychoacoustic practitioners who do their research in the literature without discriminating between accurate and faulty material. The key *fact* about present day psychoacoustics is the crying need to go carefully through what is *believed* to be true and resubmit it to *properly* performed tests. Even a casual perusal of contemporary psychoacoustic literature reveals truly naive testing procedures devoid of even the simplest controls over polarities, phase, electronic equipment stability, aligned sources, etc. For instance, one paper cited subjective approval of a control room using grossly misaligned monitors. One paper showed an ETC of a control room with a fundamental flaw of a huge console reflection and labeled it a well thought of LEDE™ room.

Time For Change

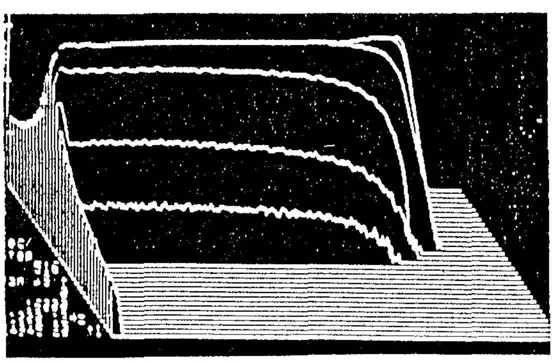
We notice that others are even less restrained than we are and some editorials (i.e., July and November editorials in "Sound & Video Contractors" magazine) must have been originally written on asbestos. There is no question that the technical sessions have become ego trips for political friends and that the exhibitors are viewed primarily as revenue generating pawns there to support an increasingly large bureaucratic structure, both actual and planned for the future.

We can always hope for internal reform but are not optimistic about it actually happening.

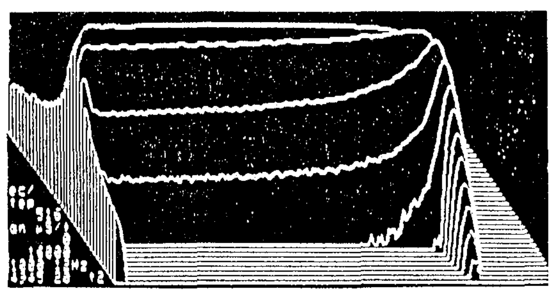
The following are excerpts from Dr. Lagadec's and Dr. Stockham's Paris AES paper, pp. 2097, entitled "Dispersive Models for A-to-D and D-to-A Conversion Systems":

> Remarkable figures, in particular, were found when analyzing cascades of A-to-D and D-to-A conversion systems, in which the components due to ripple, roll-off and phase distortion soon become relatively large. These results may provide a clue to the fact, frequently described by audio professionals with trained hearing, that digital audio (despite evident technical advantages relating to, for example, signal-to-noise ratio) has some undesirable audible characteristics.....
>
> In early, unpublished experiments, which were reported orally to the AES Technical Committee on Digital Audio's Working Group on Measurement Techniques by one of the authors (Prof. Stockham), the effect of very sharp cut-off filters on audio signals was investigated. A general qualitative result was that sharp filter roll-off leads to markedly perceptible effects, which cannot be described in the terminology and phenomenology of conventional, gentle lowpass filters.....

Editor's Note: See our measurements made in 1982 and reproduced here. These measurements were made of the Professional Sony Digital Tape Recorder during the November 1982 TEF® Workshop.



A 3-D "front view" of the above device's response. Note that it has an inherent internal delay of approximately 8-9 msec and a very sharp anti-aliasing filter.

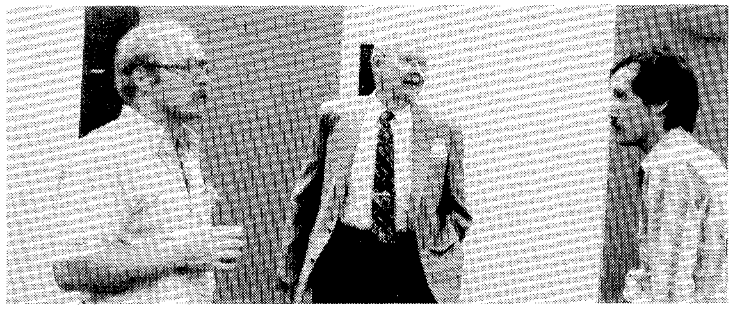


The 3-D "rear view" of the same device showing the "ringing" of the very sharp low pass anti-aliasing filter. This otherwise very fine device emphasizes the dilemma faced by those designers who insist on low sampling rates for economical reasons. Can a conjugate device be constructed? Time will tell. ♦

THERMAL SHUNT?

Bill Armstrong, a retired chemical engineer living in Florida, came to the Orlando class as a result of reading Glen Ballou's article in "Audio" that mentioned Syn-Aud-Con.

Mr. Armstrong showed his remarkably quick mental scan ability when we discussed, at the luncheon table, a telephone call we had just received asking what a "thermal shunt" was. It seems a graduate importing loudspeakers from Israel was being asked to provide them with thermal shunts. Bill thought a few minutes and then said, "It sounds like a fuse to me." Bravo! ♦



Michael Newsome - Bill Armstrong - George Bish

AES JOURNAL POLICY STATEMENT

The September issue of the "Journal of the AES" contained an interesting "Editorial Note." The discussion centered around an ad in the "Journal" by AKG in 1981-82 which stated, "Allowable sound pressure levels of 142 dB (162 dB with selectable attenuation) and the lowest self-noise figure of any microphone we know."

Steve Temmer of Gotham Audio took exception to the ad and after a lot of back and forth canting and re-canting, the AES made this statement:

> "For the future, the AES will call for a qualified independent evaluation of the veracity of claims appearing in an advertisement which has been challenged."

If the "Journal" readers want to, they can keep AES's "qualified independent evaluation" lab busy. In the same issue an ad appeared for a loudspeaker with these statements:

> "________ is the first new technology for commercial speakers in over 30 years. Each speaker is a full-range system which is highly accurate from 40 Hz to 19 kHz ± 3 dB. It radiates from the front and the back equally, and is essentially spherical across the entire frequency spectrum."

This is not to say that _______ is a good (or bad) speaker system, but I'd like to see proof that (1) it represents the first new technology for commercial speakers in over 30 years and (2) that it is essentially spherical across the entire frequency spectrum. ♦

HARD-TO-FIND DECADE BOX

Anyone working with *circuits* of any kind has desired to have variable decade boxes for resistance, capacitance and inductance. The first two are not all that difficult to find. Variable precision inductance can be another matter. Syn-Aud-Con rarely does circuit work, the single exception being our analysis of various types of equalizing filters.

We recently found a source for such decades from: IET Labs, Inc. - 534 Main Street - Westbury, NY 11590
Telephone: (516) 334-5959

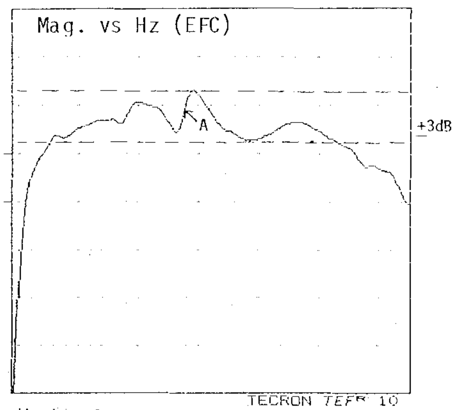
Their RCS-502 precision resistance capacitance decade and the LS-400 wide range inductance give the circuit designer:

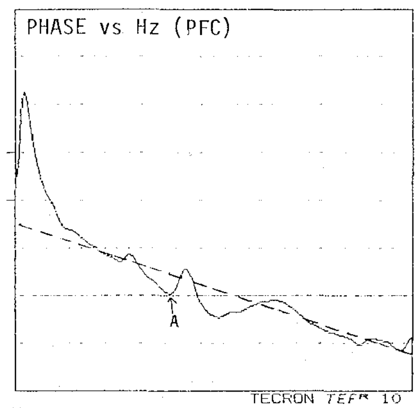
Resistance Range:	1 to 9,999,999Ω
Capacitance Range:	100 pF to 99.9999 uF
Inductance Range:	1 mH to 9.999 H

Prices we have seen are very reasonable but we're not sure the ones we have are current so we'd suggest you contact IET directly for pricing information. ♦

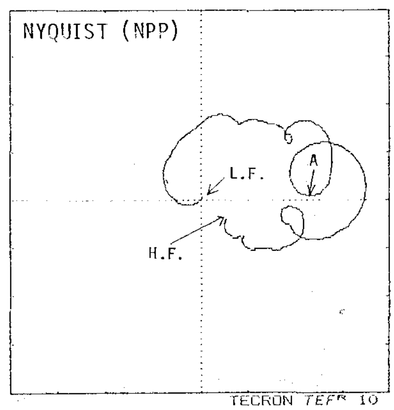
THE TOA M8230 "MEETING AMP"

The small self-powered 4-speaker TOA sound column housed in the grey metal case and used for voice reinforcement by Syn-Aud-Con for the past year is shown in the following measurements.





NYQUIST PLOT: Epycycles caused by four speakers in a column being slightly out of time with each other.



Vertical:
6dB/div with base of display at 63.2dB
0dB is located at .00002 Pa.

Vertical:
45 degrees/div.
0 degrees is at the dashed horizontal line.

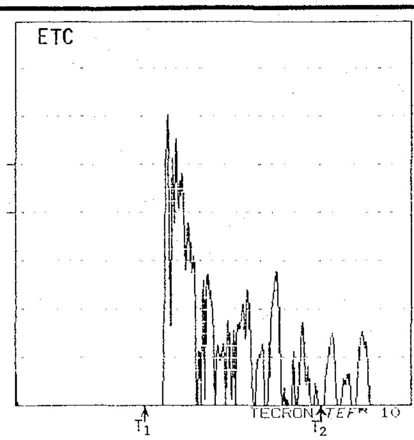
Horizontal: Auto 0.00Hz to 4998.46Hz
scale: 1366.63Hz/inch or 538.04Hz/cm.

Resolution: 4.5194E+00 Feet & 2.5048E+02Hz

Time of test: 2431 microseconds, 2.7523E+00 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 2.0000E+01Hz

Frequency range: Auto 0.00Hz to 4998.46Hz



The energy frequency curve (EFC) display revealed a very smooth (±3dB) telephone frequency response. This is perhaps almost an ideal frequency range for use with live talkers in small (60 ft. x 30 ft.) hotel meeting rooms. Because we use the HME wireless microphone system with a PZM lavalier, we get the full benefit of this loudspeaker's smoothness.

Vertical: 6dB/div with base of display at 57.2dB
0dB is located at .00002 Pa.

Horizontal: 0 microseconds or 0 Feet to
6300 microseconds or 7.13103 Feet
scale: 1.9497E+00 Feet/inch or 7.6760E-01 Feet/cm.
1722 microseconds/inch or 678 microseconds/cm.

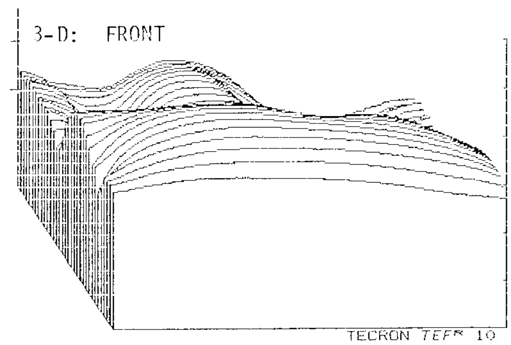
Line Spacing: 15.7882 microseconds or 1.78723E-2 Feet
Line Width: 21.472 microseconds or 2.43063E-2 Feet

Sweep rate: 5009.55Hz/Sec

Sweep range: -31668.60Hz to 31669.70Hz

Continued next page.....

The Phase vs Frequency and Nyquist plots reveal an extremely well behaved system (remember this unit has its own amplifier as well).



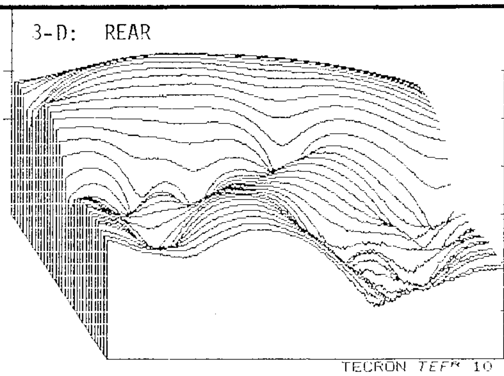
Vertical: 12dB/div with base of display at 51.2dB
0dB is located at .00002 Pa.

Horizontal: Auto 0.00Hz to 4998.46Hz
scale: 1366.63Hz/inch or 538.04Hz/cm.

Resolution: 5.6492E-01 Feet & 2.0038E+03Hz

Time of test: 2052 microseconds 2.3234E+00 Feet (front)
to 4294 microseconds 4.8613E+00 Feet (back)
-72 microseconds/step or -.08186659375 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 2.5000E+00Hz



Vertical: 12dB/div with base of display at 51.2dB
0dB is located at .00002 Pa.

Horizontal: Auto 0.00Hz to 4998.46Hz
scale: 1366.63Hz/inch or 538.04Hz/cm.

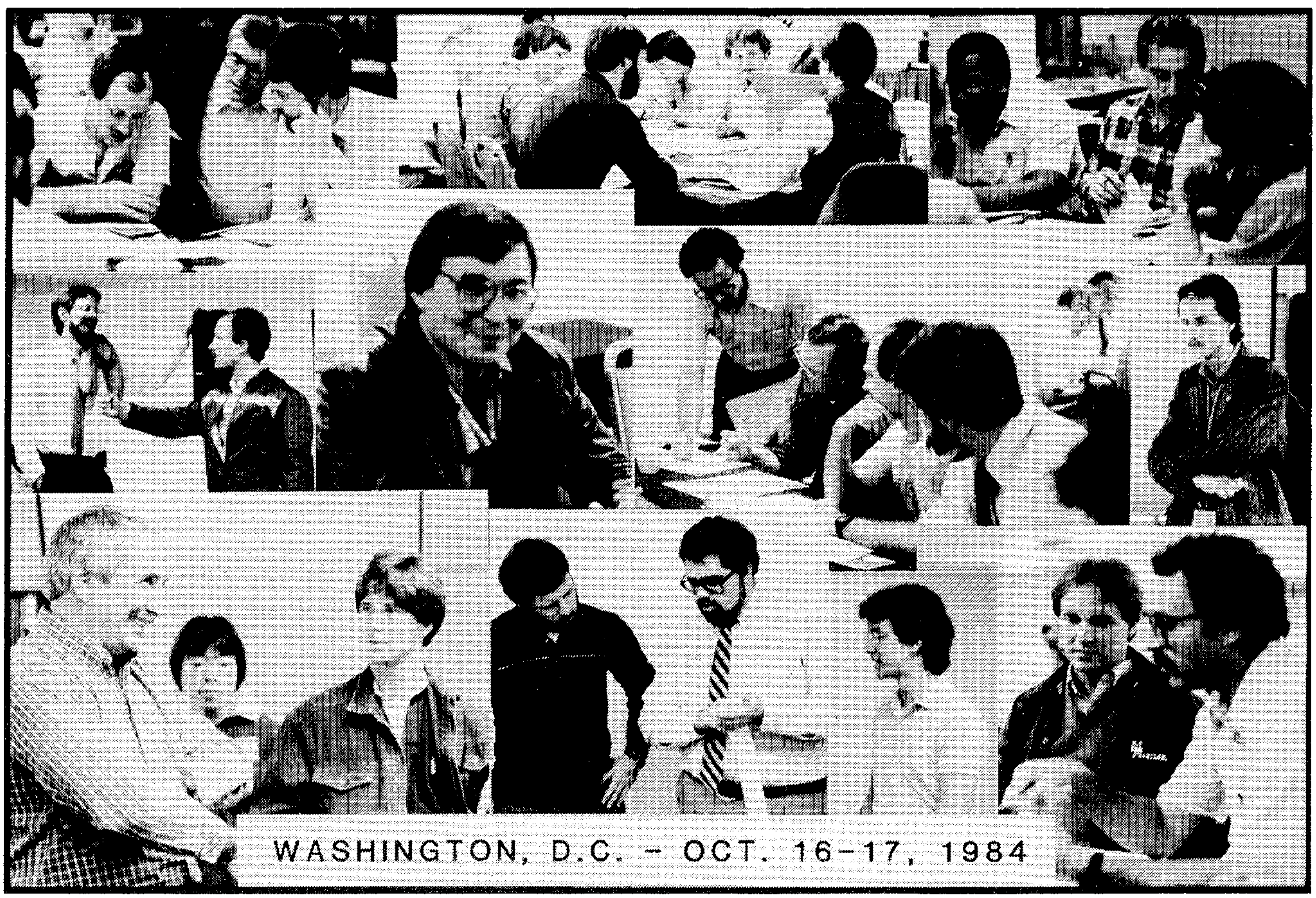
Resolution: 5.6492E-01 Feet & 2.0038E+03Hz

Time of test: 4294 microseconds 4.8613E+00 Feet (front)
to 2052 microseconds 2.3234E+00 Feet (back)
-72 microseconds/step or -.08186659375 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 2.5000E+00Hz

What was of most interest to us is the initial response in time revealing that the slight peaks and dips we observed in the EFC are the result of slightly later returns from the other loudspeakers in the column.

There are so many poor quality all-in-one meeting systems on the market that it is a genuine pleasure to find one that both measures well and sounds good. ♦



HOW TO READ THE VU LEVEL ON A VI INSTRUMENT

A volume indicating (VI) instrument is used to measure the *level* of a signal in volume units (VU). In calibration: 0 VU = 0 dBm and a 1.0 VU increment is identical to a 1.0 dB increment.

The true level reading in VU is found by:

$$\text{True VU level} = \overbrace{\text{instrument indication} + \text{attenuator or sensitivity indicator}}^{\text{Apparent Level}} + 10 \log \left[\frac{600}{Z_{actual}}\right]$$

Thus, we can have:

1. A reading from the face of the instrument (zero preferred)
2. The reading from the attenuator or other sensitivity adjustment - normally a minimum of +4 or higher. When instrument indicates zero, then the apparent level is the attenuator setting.
3. The correction factor for impedance other than the reference impedance. 600 ohms is the normal impedance chosen for a reference, but any value can be used so long as the voltage across it results in a 0.001 watt.

Calibrating A VI Instrument

The instrument should be calibrated to read a *true* level of zero VU when an input of 1000 Hz steady state sine wave signal of zero dBm (0.001 watt) is connected to it.

For example, typical calibration is when the instrument indicates -4, the attenuator value is +4 and it is connected across a 600 ohm circuit. Levels read on a VI instrument in VU when the source is the aforementioned sine wave signal *should be stated as a dBm level*.

Reading A VI Instrument On Program Material

Because of the ballistic properties of VI instruments, they exhibit what has been called "instrument lag." On short duration peak levels, they will "lag" by approximately 10 dB. Stated another way, if we read a true VU level of +8 VU on a speech signal, then the level in dBm becomes +18 dBm.

Levels stated in VU are assumed to be program material and levels stated in dBm are assumed to be steady state sine wave. This means that the associated amplification equipment when fed a true VU level of +8 VU must have a steady state sine wave capability of +18 dBm in order to avoid overload.

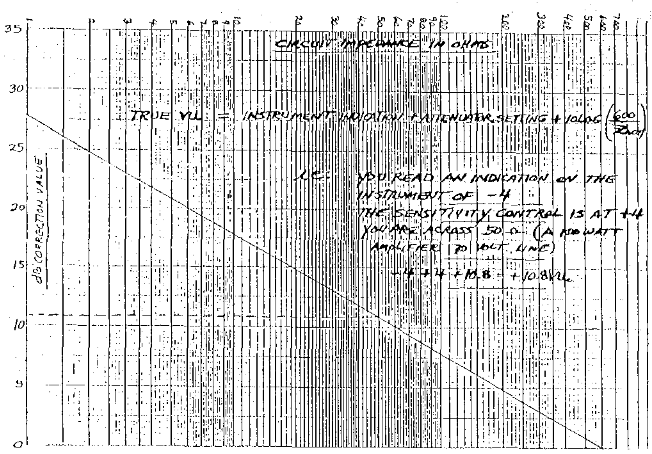
Reading Apparent VU Levels

VI instruments can be usefully employed to read apparent or relative levels. If the user has preknowledge that overload occurs at some *apparent* level, he or she can use that reading as a satisfactory guide to the system's operation though ignorant of the true level. When adjusting levels using the instrument to read the relative change in level, such as turning the system down 6 dB, it is not necessary to do so in true level readings. Instrument indication serves effectively in such cases.

Conclusion

When being given a *level*, be sure to ascertain:

1. Is it an instrument indication?
2. Is it an apparent level?
3. Is it a true level?
4. Is it a relative level?
5. Is it a calibration level?
6. Is it a program level?
7. Is it none of the above, but simply an arbitrary *meter* reading? ♦



IMPULSE vs STEADY-STATE NOISE

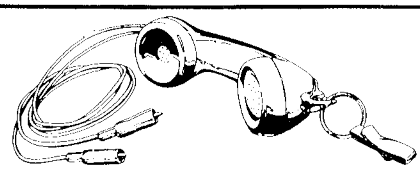
From the "Journal, Acoustical Society of America," Volume 76, No. 4, October 1984: "Effect of the temporal pattern of a given noise dose on TTS in guinea pigs" --

The noise dose necessary to obtain a given level of TTS_{15min} *is only half as great for impulse noise as for continuous noise.* ♦

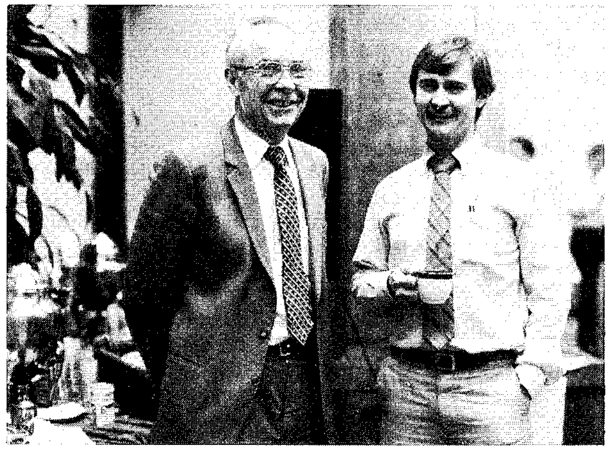
MUSIC SUPPLY COMPANY IN DALLAS

W. E. Brown, Jr., President of Music Supply Company, Inc., in Dallas, Texas, wrote us recently saying that he had received a U.L. listing on his speaker support truss and the new speaker enclosure that goes with it.

Mr. Brown is the builder of the universal test handset we show in our classes (sold by J. W. Davis & Company also). The picture shows him on the left with Rod Harvey of Comcast during the 1983 class in Dallas. ♦



TESTSET MODEL TS-1 FOR THE AUDIO SERVICEMAN



A GOOD TOURING SYSTEM

FIGURE 1a and 1b: The Nyquist displays at the right show a loudspeaker system before (Fig. 1a) and after (Fig. 1b) Sonex was applied in a crude way to cover the edge around the horn.

FIGURE 2 (below) shows a good ETC. Note that the reflections from the floor and ceiling have much the same shape as direct sound. This is typical and one of the best arguments for signal alignment.

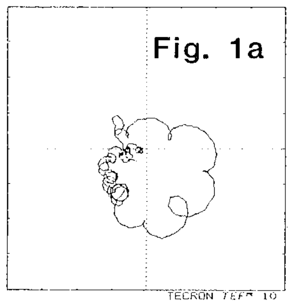


Fig. 1a

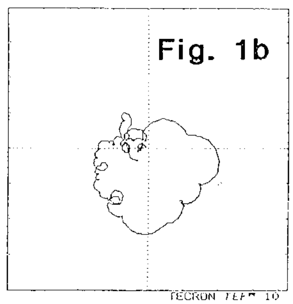


Fig. 1b

Resolution: 4.5114E+00 Feet & 2.5048E+02Hz

12dB of automatic screen gain.

Frequency range: Auto 0.00Hz to 19998.10Hz

Time of test: 7946 microseconds, 8.9785E+00 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 2.0000E+01Hz

Vertical: 6dB/div with base of display at 30.1dB
0dB is located at .00002 Pascals

Horizontal: 6300 microseconds or 7.119 Feet to
18927 microseconds or 21.387 Feet
scale: 3.9010E+00 Feet/inch or 1.5358E+00 Feet/cm.
3452 microseconds/inch or 1359 microseconds/cm.

Line Spacing: 31.6455 microseconds or 3.57594E-2 Feet
Line Width: 43.0379 microseconds or 4.86328E-2 Feet

Sweep rate: 5009.55Hz/Sec

Sweep range: 99.58Hz to 31699.70Hz

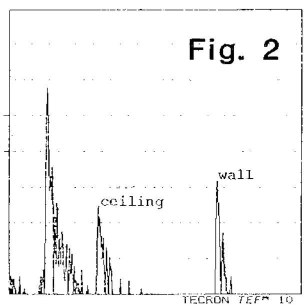


Fig. 2

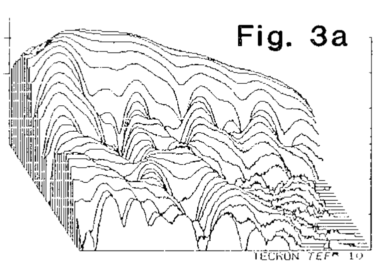


Fig. 3a

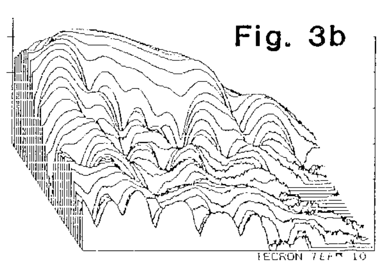


Fig. 3b

Vertical: 12dB/div with base of display at 30.1dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 19998.10Hz
scale: 5467.68Hz/inch or 2152.63Hz/cm.

Resolution: 3.3835E-01 Feet & 3.3397E+03Hz

Time of test: 9908 microseconds 1.1196E+01 Feet (front)
to 7724 microseconds 8.7282E+00 Feet (back)
-70 microseconds/step or -7.959352066532E-2 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 1.5000E+00Hz

FIGURES 3a and 3b (right): Waterfall (backside -- that is, how the speaker shuts off) of two systems. Fig. 3a is of the speakers about 1 ft. apart; Fig. 3b is with the edges touching. That's 12 dB more loss of high frequencies in 3b than in 3a. Incidentally, the measurement on the driver is practically flat *before* mounting to the horn. The mounting to the horn and driver caused about 12+ dB loss in Fig. 3a. ♦

"POWER OF LIMITS"

While in Nashville for the LEDE™ Workshop, Steve Blake, of Professional Recording & Sound in Boston, presented us with a copy of *The Power of Limits* by György Docozi. This book contains gems like "The greatness of teachers is not measured by how much they know but by how much they share" (The Rev. Jesse Jackson) and "Enthusiasm means literally 'God within'."

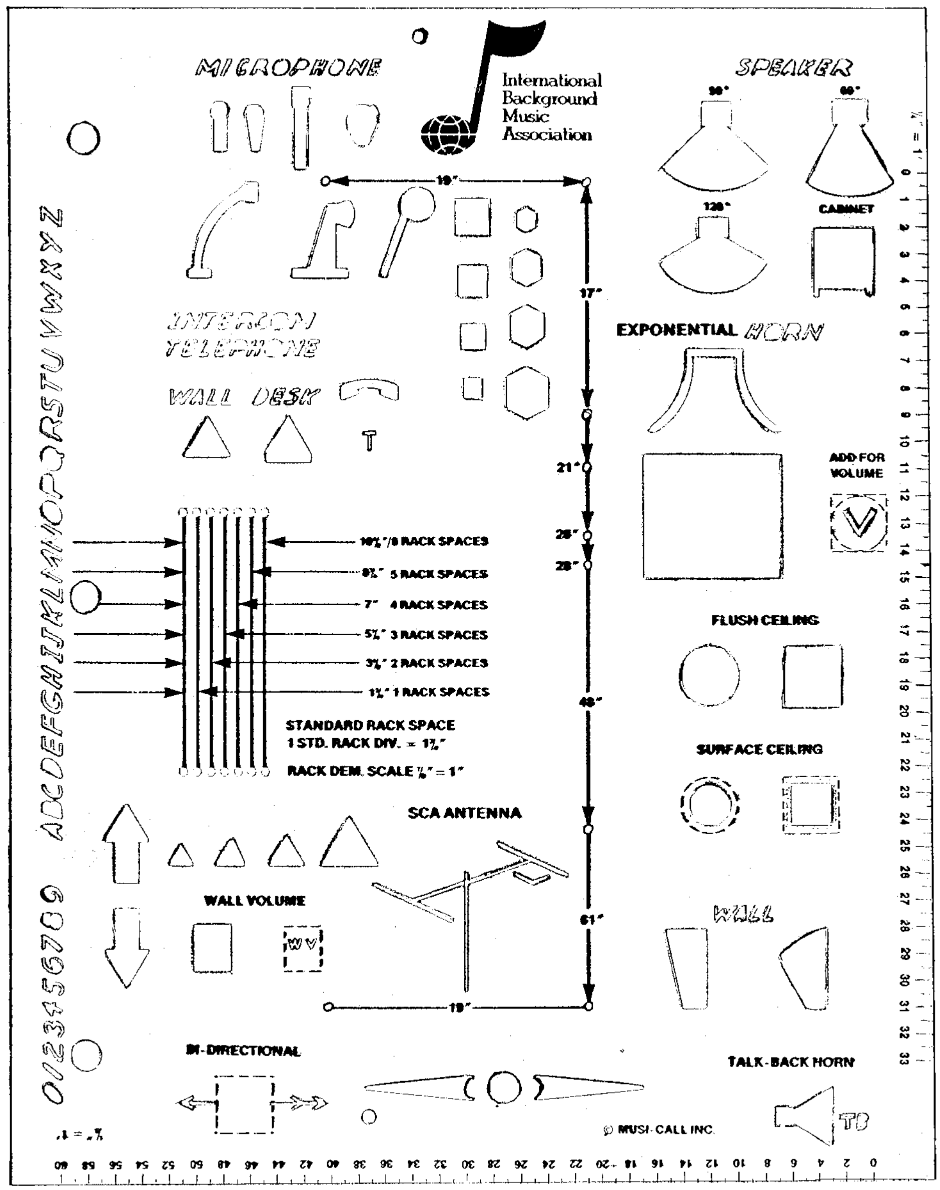
The author's definition of wisdom and knowledge bears serious consideration. "Among the essentials needed to turn mere survival into the art of living, perhaps none are more important than wisdom and knowledge. In a certain sense, these two human attributes are almost undistinguishable from each other; in another sense they are polar opposites. Wisdom is a putting together, knowledge a taking apart. Wisdom synthesizes and integrates, knowledge analyzes and differentiates. Wisdom sees only with the eyes of the mind; it envisions relationship, wholeness, unity. Knowledge accepts only that which can be verified by the senses; it grasps only the specific and the diverse."

One final quote as it expresses the American contribution to world thought:

"Thomas Jefferson realized that all human beings 'are endowed by their creator' with equal rights to 'life, liberty, and the pursuit of happiness' because none are born 'with saddles on their backs, nor a favored few booted and spurred'."

It's that thought rather than any material thing that has brought the world's free men to our shores for the past centuries. ♦

SOUND SYSTEM TEMPLATE



We are reproducing the best photocopy we were able to make of the new sound system design template produced for the International Background Music Association

Dan Lee

Musi-Call, Inc.
4949 West Belmont
Chicago, IL 60641
(312)685-7850

recently sent us one to look at and we are most impressed with its usefulness. Proposals minus drawings are an easy way to lose a bid when your competitor supplies them. This new 8-1/2" x 11" template makes such quick drawings easy as well as neat. ♦

SHURE FB31 PORTABLE MIXER

Since first writing about the new Shure FP31 portable mixer, we have acquired one and are using it as part of our test equipment setup. This is truly a superb unit and the combination of a totally versatile mixer and battery operation is enough to spoil any hardened audio man.

The instruction manual that comes with it is a little textbook and our sole objection is their continued use of phase when they mean polarity and OVU = +4dBm when they mean "for an instrument indication of '0' the output is +4VU if a complex program wave or, if a sine wave, then +4dBm."

The small size belies the big precision built into this instrument. We now use it as part of every microphone test we make. The three 9 volt batteries insure accessibility to new ones at any drug store. The balanced 600 ohm transformer coupled inputs allow connection to anything from "Mother Bell" to the largest auditorium system. We even use this unit as the most precise way to feed a 150Ω mic line from the stage to the console from our unbalanced 600Ω random noise generator. One input has noise on it and the second input allows talk back to the operator up at the console (or you can use the slate microphone built into the instrument).

If I were in the rental business, this would be a workhorse mixer for use with any of the "meeting room" amplifier-speaker setups we encounter, especially when you consider that we have yet to see one with a front end anywhere close to the quality and performance of this unit. ♦

TESTED "FOR SALE" EQUIPMENT

In the "Classified" section of this Newsletter is a list of test equipment being offered by Richard C. Cabot, formerly with Tektronix and now an independent consulting engineer. He recently, at my request, sent me data on the B & K peak, RMS, average meter included in the list. I have excerpted the data from his letter to show how skilled engineers test the validity of specifications.

I used a tone burst generator set to deliver a single cycle burst at the press of a button. I set the meter to read the positive peak of the waveform. Compared to the steady state peak value, I got the following readings:

1 cycle @ 5 kHz	within 0.25 dB
1 cycle @ 10 kHz	within 1 dB
1 cycle @ 20 kHz	within 3 dB

If the meter is set to read both positive and negative peaks, i.e., full wave rectification before peak holding, the readings were as follows:

1 cycle @ 10 kHz	within 0.25 dB
1 cycle @ 20 kHz	within 1 dB

The meter is still in the B & K catalog, so you should be able to get more information from them as to guaranteed specs. ♦

CLASSIFIED

FOR SALE: Used test equipment -

B&K 2305 Level Recorder	$1000.00
B&K 1022 Beat Frequency Oscillator	$1000.00
B&K 2425 Peak/Average/RMS Voltmeter	$ 750.00
B&K 2308 Measuring Amplifier	$ 850.00
B&K 4220 Pistonphone	$ 400.00
B&K 4290 Accelerometer Calibrator	$ 500.00
Ivie IE10A Octave Band RTA	$ 400.00

Contact: Richard C. Cabot, 12820 S.W. Washington St., Beaverton, OR 97005
Telephone - (503) 644-9727

FOR SALE: GenRad 2512 FFT - excellent condition. _NO_ reasonable offer refused.

Contact: Don Davis, Syn-Aud-Con, P. O. Box 669, San Juan Capistrano, CA 92693
Telephone: (714) 728-0245

INDUSTRIAL RESEARCH PRODUCTS, INC.
A Knowles COMPANY

SHURE®

SYN-AUD-CON SPONSORS

HM ELECTRONICS, INC.

TOA ELECTRONICS, INC.

Syn-Aud-Con receives tangible support from the audio industry, and eleven 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.

Their presence on this list as a Syn-Aud-Con sponsor indicates their desire to work cooperatively with you in professional sound.

Bose Corporation
Community Light & Sound, Inc.
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