SYNERGETIC AUDIO CONCEPTS

SYN AUD CON

newsletter

VOLUME 11, NUMBER 3
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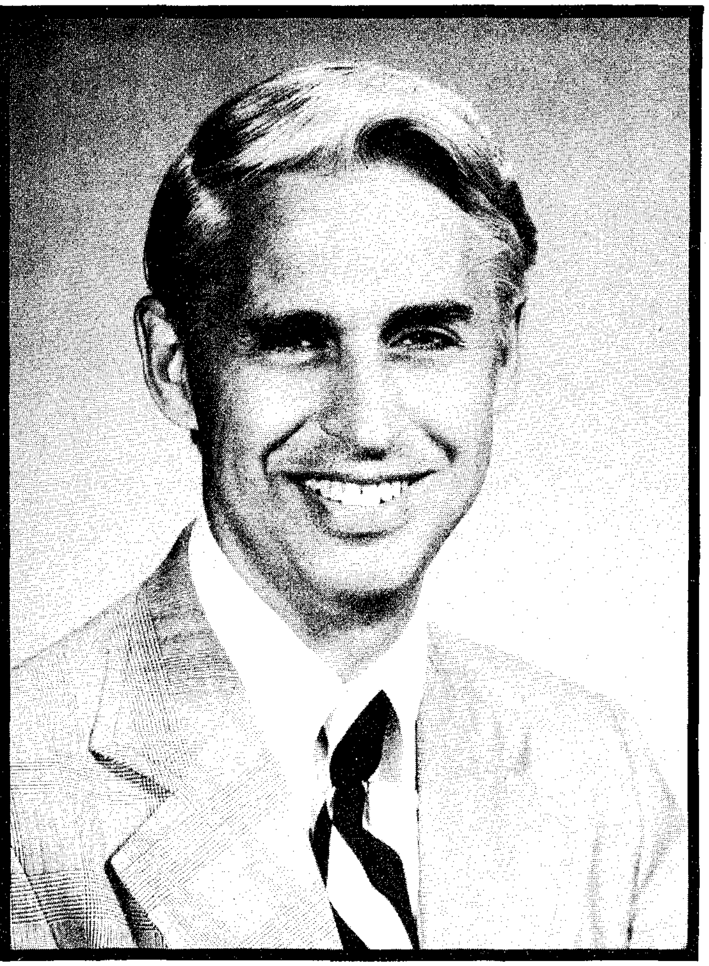
SYNERGETIC
Working together; co-operating, co-operative

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

S. N. SHURE



J. H. KOGEN

SHURE® – A SYMBOL OF INTEGRITY

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THE FRONT COVER

It is significant that the first personalities ever to appear on the cover of a Syn-Aud-Con Newsletter should be S. N. Shure and Jim Kogen of Shure®.

Mr. Shure is a genuine legend in his own time. Like the prophets of the Old Testament, his major influence is mental, not physical, and manifests itself in the quality of men surrounding him. Read any technical paper ever given by Shure personnel and you'll discover that the work was significant, the techniques used the best engineering practices of the day, and those of us in the audio education business can use them as textbooks on the subject covered.

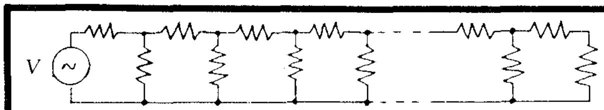
Jim Kogen, the President of Shure, has participated in many such projects and can truly be called a thoroughly professional engineer. One of his remarkable traits is a keen awareness of not knowing everything when acclaimed as a genuine authority in his own fields of interest.

What does all this add up to? First of all, it would be presumptious to attempt to summarize on an editorial page accomplishments of men this talented and prolific in the audio arts and sciences. We'd just like to express our pleasure in observing that audio's most successful and senior pioneer company, still headed by its founder, has reached such prominence in our industry with true gentlemen at the helm, high quality products still in their inventory, and a reputation for integrity not likely to be matched in our industry.

Yes! Good guys do win, and win big. ♦

A CIRCUIT CURIOSITY

Who can write the correct equation to obtain the Fibonacci numbers via Ohms Law?

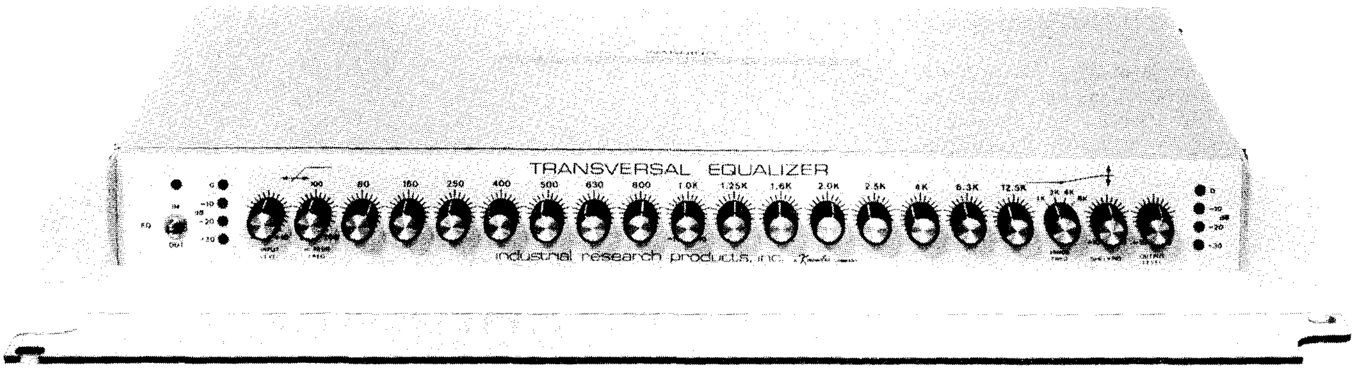


20th century version of Ahmes' method. The calculation of the currents due to a known voltage V in the above network becomes very cumbersome when the circuit is worked from left to right. So one assumes a (probably wrong) current of 1 ampere in the last branch and works the circuit from right to left (which is much easier); this ends up with the wrong voltage, which is then corrected and all currents are scaled in proportion. Just as was done by the Egyptians in 2,000 B.C.

The above circuit is also a mathematical curiosity for another reason. If all its elements are 1 ohm resistors, and the current in the last branch is 1 ampere, then the voltages across the resistors (from right to left) are Fibonacci numbers (1, 1, 2, 3, 5, 8, 13, 21, 34, 55, ... each new number being the sum of the last two).

A BETTER WAY TO BUILD AN EQUALIZER

Industrial Research Products, Inc., IRPI, is again living up to the "Research" in their name. First, they developed the best and most reliable digital signal delay available in our business. They followed this with the most versatile automatic mixer available. Now they've outstripped even their own past accomplishments with a Transversal Filter Equalizer.



Transversal filters were first described, to our best knowledge, by H. E. Kallmann in the July 1940, Volume 28 issue of the "Proceedings of the IRE." That's 44 years ago and the IRPI equalizer is the first audio product to solve the very real problems of converting theory into practice.

Dick Heyser has discussed "Kallmann" filters in his papers and, indeed, the list of those familiar to Syn-Aud-Con that are aware of what this means are those of Heyser's stature.

Another quality of IRPI that proceeds directly from the leadership of men like Hugh Knowles and Mahlon Burkhard is the direct, accurate and dependable technical descriptions of what they have done. Their reasoning on what the advantages of this new equalizer does is reproduced above.

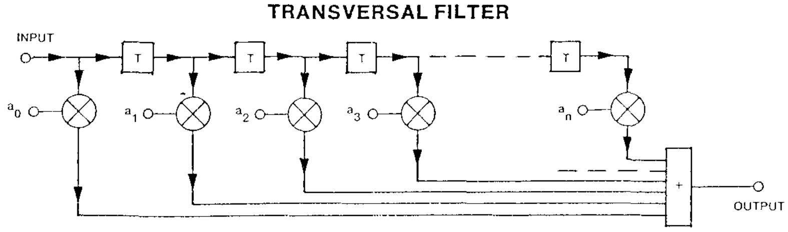
WHY A TRANSVERSAL EQUALIZER?

Traditional equalizers assembled with individual tuned filters produce frequency response ripples because of inexact combining of adjacent filters. While these responses may be of small direct importance for sound quality, their presence reflects an extraneous transient ringing which may compromise sound quality. Ripples as small as 1.0 dB in the combined response could produce transient ringing errors only 20 dB below the desired signal.

To suppress this ringing, better sounding equalizers frequently employ low Q filters, reasoning that if each filter rings less, the combined transient response will be free of extraneous ringing. While reducing the ringing, highly interactive controls result that require an exasperatingly long time to adjust to a desired system response.

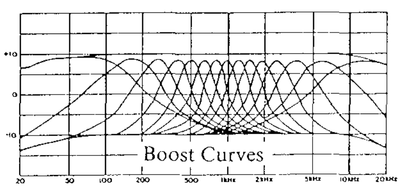
The transversal filter solves this problem fundamentally and structurally. The transient response of the DG-4017 *is not* the result of a sum of individual transient responses but is the weighted response of a single tapped delay chain. (See Transversal Filter Insert). Thus the transient response is synthesized from a single fixed response circuit. The resulting ideal transient response is reflected in the small ripple response specifications of the DG-4017.

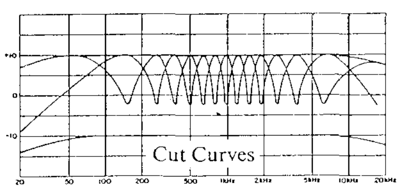
TRANSVERSAL FILTER

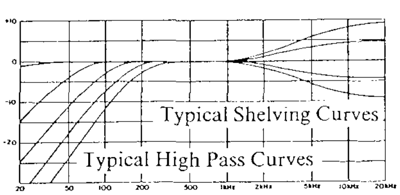


The input signal is delayed by a chain of N stages, each stage delaying the signal by amount T. Each stage output is multiplied by a weighting factor a_n and summed together for the final output. The frequency response of the filter is determined by the relative values of the weighting factors, a_n.

DG-4017
FREQUENCY RESPONSE FAMILIES







ARCHITECTS SPECIFICATION

The equalizer frequency response ripple shall be less than ± 0.1 dB over 100% of a 20 dB adjustment range. The equalizer shall function equally well in a cut only, boost only or in a boost/cut mode of adjustment. The equalizer shall use non-resonant circuits to achieve the equalizing function to avoid unnecessary ringing in response to a transient signal. A Butterworth 12 dB/Oct high pass filter, continuously adjustable from 20 Hz to 200 Hz, shall be provided. A high frequency shelving filter with hinge frequencies of 1 kHz, 2 kHz, 4 kHz and 8 kHz and with a continuously adjustable boost or cut slope and with a maximum end range of ±10 dB, shall be provided. Both input and output levels shall be adjustable and monitored independently by LED arrays. Both input and output shall be transformer coupled. THD shall be less than .1% at 1 kHz and +4 dBm. Noise shall be 100 dB below full output with controls set for unity gain. Full output shall be +23 dBm into 600 Ω. A bypass switch with flashing LED reminder shall be provided. The equalizer shall operate from 50/60 Hz with 115/230 VAC. The equalizer shall be the Industrial Research Products, Inc. DG-4017 Transversal Equalizer.

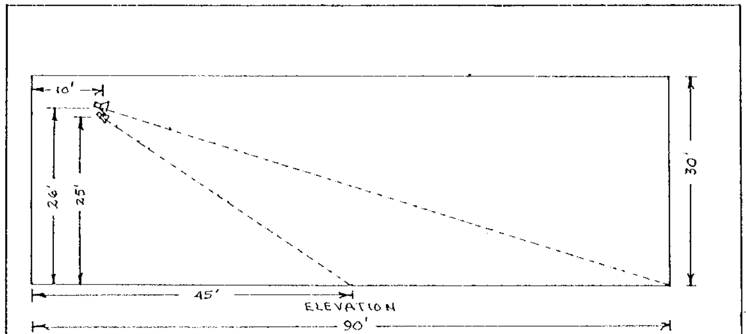
There is also a version that plugs into their automatic mixer (the Transversal Equalizer Module DE-208).

In Syn-Aud-Con's opinion, this is the first genuine step forward in equalizers since the work done in designing the original 1/3 octave units done by ourselves. IRPI's equalizer is absolutely a major step forward and should quickly prove to be the optimum choice available in the industry. ♦

LOUDSPEAKER COVERAGE PLOTTING

Our audio crystal ball has just disgorged a future event. Gene Patronis has a talented candidate for a Ph.D. working on a TEF® analyzer program for loudspeaker coverage plotting using the exceptional graphics capability of this analyzer as a computer.

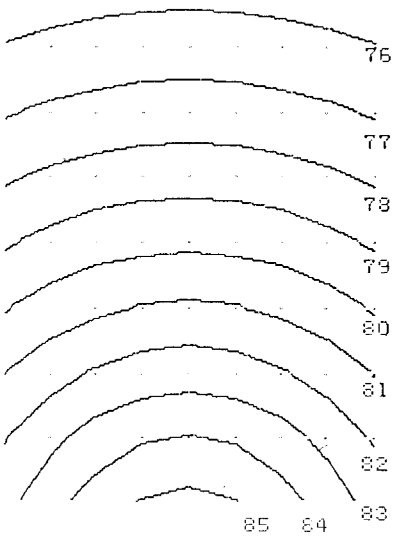
We asked Gene to plot a "long throw" and a "short throw" loudspeaker in a small meeting room. He quickly returned to us the following plots from 100 Hz to 2000 Hz as printed by a printer attached to the TEF analyzer.

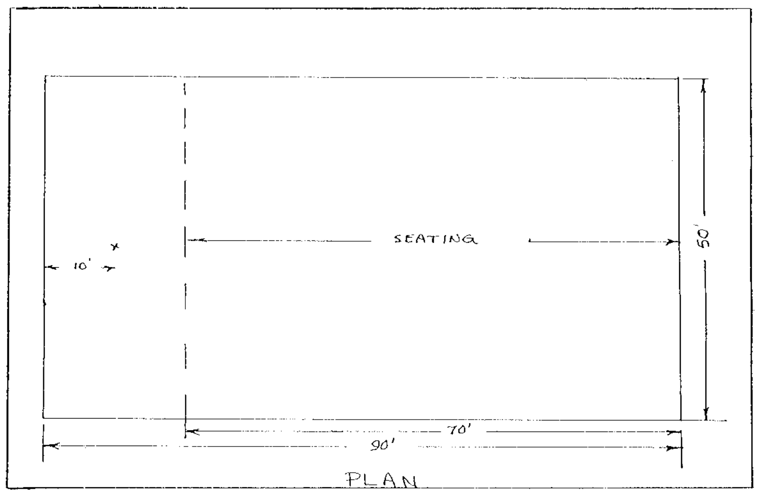


NOTE: LOUDSPEAKERS ARE 8" EXTENDED RANGE WITH A SENSITIVITY OF 96 dB/WATT (4', EACH LOUDSPEAKER IS ENERGIZED WITH 1 WATT OF DRIVE. ACTUAL LOUDSPEAKER DIRECTIVITY FUNCTIONS ARE EMPLOYED IN THE CALCULATIONS.

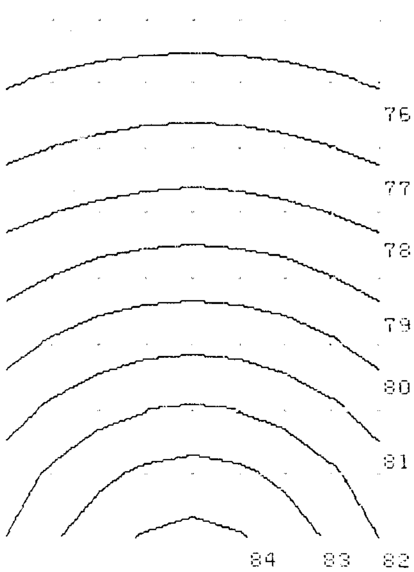
TECRON *TEF*™ 10

100 HZ
MAX = 85
MIN = 76
INTERVAL = 1DB

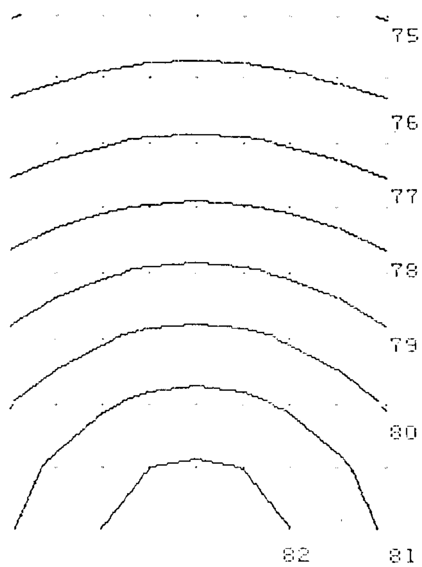




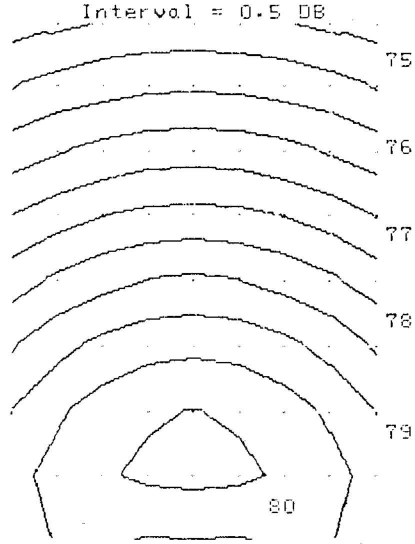
200 HZ
MAX = 84
MIN = 76
INTERVAL = 1 DB



300 HZ
MAX = 82
MIN = 75
INTERVAL 1 DB



1 400 HZ
Max = 80
Min = 75
Interval = 0.5 DB



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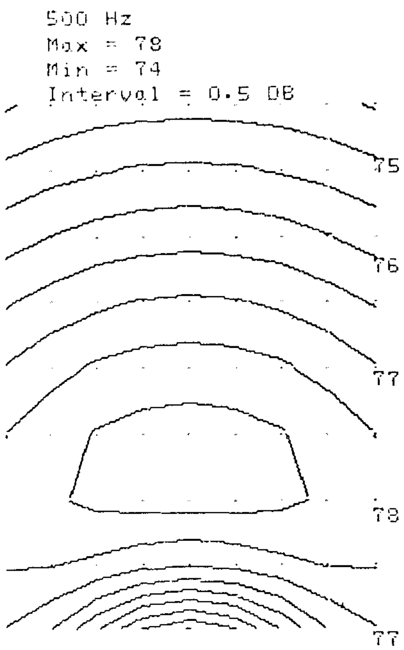

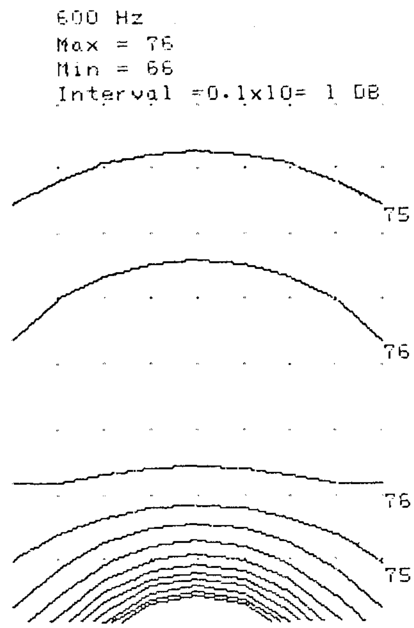

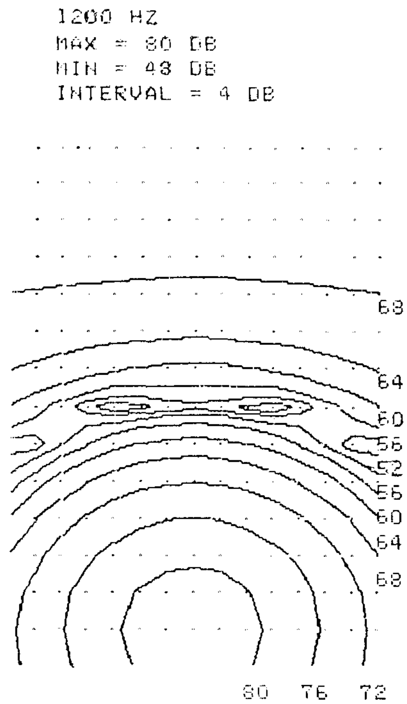

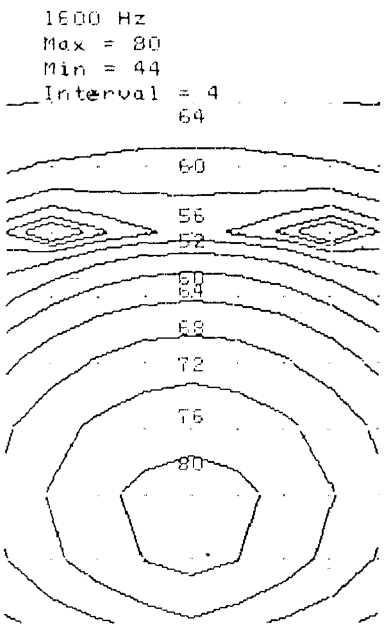

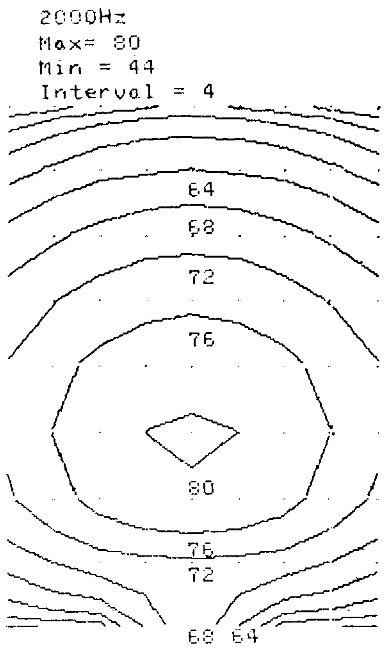

This will be one of the working tools at the upcoming Loudspeaker Array Workshop in Nashville, Tennessee, this spring. ♦

LOUDSPEAKER ARRAY
AND
SOUND SYSTEM DESIGN
WORKSHOP

June 19-21, 1984

Marriott Hotel
I-40 & Briley Parkway
Nashville, TN

LEDE™ WORKSHOP

Where: Acorn Sound Recorders
(Oak Ridge Boys new studio designed by Robert Todrank of Valley Audio.)

When: September 11 - 13, 1984

Who:

Workshop Chairman:	Neil Muncy
Workshop Host:	Robert Todrank
Intructor:	Dr. Peter D'Antonio
Guests:	Chips Davis
	Russ Berger
	Glenn Meeks

Fee: $600.00

Sponsored By: Syn-Aud-Con

LOUDSPEAKER ARRAY and......
SOUND SYSTEM DESIGN WORKSHOP
with emphasis on computer programs

June 19-21, 1984 – Nashville, TN

WORKSHOP INSTRUCTORS

David L. Klepper, KMK Associates — Eugene T. Patronis, Ph.D., Georgia Tech

John R. Prohs & David E. Harris, Ambassador College

Imagine a computer program for sound system design that:

1. Calls up a horn and driver by name in order to automatically enter its Q, C_L, sensitivity, power handling, etc., into your calculation.
2. Contains built-in lists of absorption coefficients.
3. Contains built-in menu driven aids for finding surface areas of any shape and volume of any size and type.
4. Does the complete Community sphere program for loudspeaker coverage.
5. Contains a complete electrical gain subroutine as well as the usual acoustical gain subroutine.
6. Complete room analysis, of course.
7. The very latest Peütz $\%AL_{cons}$ equations (new as of March 1984) taken from *privately circulated* new work of V. M. A. Peütz.
8. Is considered by Syn-Aud-Con to correctly use the optimum equations for each area of inquiry.

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> E L.E.P.H.A N.T.
>
> ********************
>
> ELectroacoustical Engineering Program using Holistic ANalysis Techniques
>
> John R. Prohs
>
> David E. Harris
>
> TECHNICAL OPERATIONS AND ENGINEERING
> AMBASSADOR COLLEGE
> 300 West Green Street
> Pasadena, CA 91129
> (818) 304-6047

Well, it exists.

It will be shown and used by participants at the LOUDSPEAKER ARRAY and SOUND SYSTEM DESIGN WORKSHOP, June 19-21, 1984, at Nashville, Tennessee.

Dave Klepper has no competition for the title of the "most experienced sound system designer in the world today." Dave has already adopted the Community sphere method of loudspeaker coverage calculation and, as is typical of Dave, he will not just talk about such techniques but demonstrate to us a real life result. Dave also has used TEF® analysis to set up several jobs, including an outstanding new pew back system. We sincerely doubt that Dave has a peer when it comes to correlating the old with the new, having done the best jobs in both cases. When Dave takes part in a Syn-Aud-Con seminar, he is free of the constraints imposed upon any consultant at a manufacturer's seminar, if only out of politeness. Dave's free to tell it like it is at a Syn-Aud-Con workshop.

Gene Patronis has rapidly carved himself a major niche in our industry. Recently cited at a major manufacturer's stockholder's meeting as a person able to save the company, his expertise in engineering matters is highly regarded even by non-technical "professional" managers.

Team members of the class will design a sound system from drawings of a church. Then the third day we will go on location to the church to measure the system recently designed by David Klepper. Truly a unique opportunity.♦

REFLECTIONS AND A PIECE OF SONEX

Precedence Effect

Once in a while we hear someone say that a signal delay was used, perfectly calculated and installed, but the Precedence Effect didn't work. Late reflections could be the problem. In our Orlando class, our Precedence demonstration was not working. We suspected a reflection off the back wall. We used a sheet of Sonex behind a listener. Then the Precedence Effect was good.

Late Reflections In The Audience

A piece of Sonex is extremely effective anytime you suspect that a late reflection is causing a problem in a certain area. Talk into the sound system while someone moves a piece of Sonex around the head of a person in the problem area. Most of the time, the Sonex will intersect the late reflection and it is not difficult to locate the area of the offending surface causing the problem. Once isolated, it is easy to correct, if the owner wants to spend the money.

Reflections At The Microphone

On the other end of the sound system, if you suspect that a reflection is causing feedback into the microphone, use a small piece of Sonex moved slowly around the microphone while it is at feedback. You may be able to intersect a reflection causing a serious feedback problem. If it is a permanent installation and gain is a problem, it may be worth correcting the reflection--or moving the location of the microphone out of the path of the reflection.

Early Reflections Around The Loudspeaker

Always keep the area around the front of a loudspeaker free of hard surfaces by padding with a soft material like Sonex. The reflections may not cause feedback, but the reflections will absolutely deteriorate the quality of the sound. ♦

A NEW SYN-AUD-CON PROJECT

Syn-Aud-Con's graduate list is now over 5,000 individuals involved in audio. They would like to have equipment specifications from manufacturers that are meaningful to them in sound *system* design work. For example:

Microphone Sensitivity

The EIA specification N dBm when an acoustic Lp at the diaphragm is zero dB. (This is a calculated value from an open circuit voltage measurement.)

$$\text{EIA rating in dBm} = L_{AIP} - 94 \text{ dB}$$

where: L_{AIP} is the available input power *level* for an acoustic input of 94 dB, i.e., 1 pa.

$$L_{AIP} = 10 \text{ LOG} \left[\frac{(E_0)^2}{0.001 \, R_S} \right] - 6\ 02 \text{ dB}$$

where: E_0 is the *open circuit* voltage obtained when an Lp = 94 dB is present at the microphone diaphragm

R_S is the *rated* impedance of the microphone.

PREFERRED FORM OF SPECIFICATION

1. EIA rating
2. Rated and actual measured impedance
3. Open circuit voltage obtained at an Lp = 94 dB

Benefit: Performer's level on a sound level meter may be directly *added* to EIA sensitivity rating yielding L_{AIP} in dBm at input of *system*.

Loudspeaker Sensitivity

The EIA specification N dB at 30' with an electrical input of 0.001 watt, i.e., 0 dBm.

This measurement can be normally made at 1.0 watt (deduct 30 dB from reading) and at any desired reference distance US or SI and then converted by means of the inverse square law relationship to our desired reference distance. (If desired, 33 feet in US and 10 meter in SI, since both distances provide the desired quality of being close to critical distance, D_C, in a majority of rooms our graduates would be working in, such as, churches, gymnasiums, school auditoriums, etc.)

The power the test is to be run at is +30 dBm into the specified impedance value, i.e., a resistor, and then that level is connected to the actual device. Input signal is to be the 2000 Hz octave band for high frequency units and the 250 Hz octave band for low frequency units. When special devices are encountered, i.e., super tweeters above 5000 Hz or sub-woofers below 100 Hz, then the choice of octave band is left to the discrimination of the tester, but must be stated in the specification.

PREFERRED FORM OF SPECIFICATION

NdB at 33 feet (10 meter) from an electrical input power of 0 dBm.

Benefit: Sound system designer or installer *adds* electrical output power of amplifier in dBm to EIA sensitivity of loudspeaker - and obtains Lp at or near D_C.

Loudspeaker Directivity

Q, N, $C_\angle$, x^o, horizontal y^o, vertical Isobar plot for -3, 6, 9, 12 dB. Q is for the 2000 Hz *octave* band using pink noise as a signal source. Increments are every 10^o measurement made at 10 meters. Any level over 18 dB below on-axis is not included in measurement, i.e., weightings.

Mixers And Other Electronic Devices Preceding Final Amplifier

1. Output level is L_{AIP}
2. Maximum input level in dBm and *actual measured* input impedance
3. *Actual measured* output Z in addition to rated output Z
4. Input sensitivity for full output expressed as an input voltage

PREFERRED FORM OF SPECIFICATION

1. Output power N dBm (not into a load, but as an L_{AIP} which is not dependent upon the next device).
2. Maximum input level in dBm
3. Sensitivity - N volts for full output
4. Actual and rated input and output impedances.

Benefit: When the L_{AIP} from the previous device, i.e., microphone line arrays, etc., is known, then the desired gain value if available merely by subtracting the input L_{AIP} from the output L_{AIP}.

Continued next page...

Power Amplifiers

1. The output power in dBm
2. The actual output Z and the rated output Z
3. The input sensitivity: N volts for full output.

If it is desired to give a "Gain" figure for an electronic device, then the expected input conditions must be specified, i.e., the R_S of the device driving the power amplifier. Normally the system designer/installer will take the sensitivity figure as an E_{IN} and obtain the E_S for the preceding device. The calculation of the individual device's gains or losses falls to the designer/installer as he or she is the one that determines the actual interfaces for the system as a whole.

PREFERRED FORM OF SPECIFICATION

N dBm output power from N volts at input.	Input Z_{actual} = X	Input Z rating = y
	Output Z_{actual} = Z	Output Z rating = w

Benefits: Loudspeaker sensitivities (EIA form) can be rapidly matched to electrical power output levels. Gain of device easily found for any input device.

Syn-Aud-Con Proposal To Manufacturers

The attached forms will be used to put out a Syn-Aud-Con Tech Topic of preferred specifications from manufacturers. These are considered preferred specifications because:

1. They are easily *field* duplicatable.
2. They are the *key* parameters needed by a designer/installer.
3. Possession of them reduces in the field calculations to an absolute minimum.

While, as in all specifications, there is room to manipulate the data, it is our belief that any manufacturer who does so will be quickly detected and exposed in a marketplace having as many active TEF® analyzers as does the commercial sound business.

Key Question

The key question is how many graduates with test equipment and manufacturers interested in helping generate meaningful "system" specifications will send in the necessary test data on their products along with suggesting material regarding test technique and equipment? Syn-Aud-Con *will* publish and advocate the use of such specifications. Let us hear from you. ♦

LVW DIGITAL PROPORTIONAL AMPLIFIER

Our "grads" tell us about unique products that fill a need not previously addressed. Lee Ritterbush of LVW in Colorado Springs told about a new product--a Digital Proportional Amplifier, DPA 1624.

The first multichannel amplifier which automatically adjusts the volume level of each loudspeaker in a sound system separately and independently from the other loudspeakers.

This amplifier is designed for use in council chambers, hotel conference rooms, broadcast studios, corporate board rooms, court rooms, theatres, etc.

The amplifier is also the first fully programmable audio product which utilizes the accessory logic outputs on automatic mixers and computers.

A typical application is a sound reinforcement system where automatic proportional dimming of loudspeakers is desired. For example, in a distributed loudspeaker system where a person speaks into a microphone, the loudspeaker directly overhead is automatically fully attenuated and each loudspeaker further out from the microphone is attenuated proportionally less.

If you're interested in more information, contact LVW Electronics, 2400 Naegele Road, Colorado Springs, Colorado 80904 (303)471-8430. ♦

"MIGHTY MINI®" HAND TRUCK

The TEF® analyzer weighs 50 lbs. Mary Gruszka travels a lot in her work as a CBS engineer and she also attends almost every Syn-Aud-Con workshop (TEF, Loudspeaker Array, LEDE™, and Concert Hall Workshops in one year).

A cart that would hold together is important. We bought one at a local department store which shed parts all the way down the long terminal hallway at LAX under the weight of a TEF analyzer and our luggage.

Mary scouted out a very sturdy, high quality "hand truck" with a 5 year warranty, guaranteed to hold 300 lbs., and step sliding features.

- Comes completely assembled
- Made of chromed steel tubing
- Size fully collapsed: 7" x 16-1/2" x 24"
- Weight: 11 lbs.
- Height of cart handle fully extended to 41" high
- Platform size: 13-1/4" wide x 11" long.

Order from MCG Audio Consulting, 88 Myrtle Avenue, Edgewater, NJ 07020. Cost is $60.00 including shipping via UPS in the United States. ♦

OH! YOU MEAN SIX POINT ZERO TWO

Richard S. Schultz of Advance Sound and Electronics in Sacramento, CA, called my attention to a mistake I am prone to make.

As he pointed out to me, there is a fundamental difference between "zero" (Ø) and "oh" (0). Teachers in school do not always make the difference clear to their students. One should express 6.02 dB as "six point zero two" and not as six point "oh" two.

Carelessness in this terminology really bears fruit when you are using computers. Richard further pointed out that the same nomenclature can cause considerable confusion with parts, numbers, purchase orders, etc. ♦

"METER LAG" AND "HEADROOM"

The difference between "meter lag" and "headroom" is often a point of confusion in discussing the total electrical power required for a given acoustic level.

If we measure, with a conventional sound level meter, an L_p = 80dB, we normally add 10dB to the measured figure to account for the "meter lag" caused by the carefully designed ballistic parameters of the meter circuit.

If, in addition, we wish to have "headroom" between our "peak" level of program material and our overload or clipping point, we then add 6dB more to our original figure.

$$L_p + \text{Meter lag} + \text{Headroom} = \text{Level for which power will be planned.}$$

In our example above, we would calculate

$$((L_p = 80) + (\text{Meter lag} = 10) + (\text{Headroom} = 6)) = L_t = 96 \text{ dB}$$

We would then design our system to allow capability to produce an L_p of 96dB, but we would operate it at an L_p = 90 for peak level. Thus, we have as "headroom" between our peak level and a level that would drive the system into clipping of 6dB. When our sound level meter indicates an L_p = 80 for dynamic speech and music signals, we would know that our peak levels are at an L_p = 90dB and that our sound level meter could indicate a high as an L_p = 86dB before we hear objectionable distortion. ♦

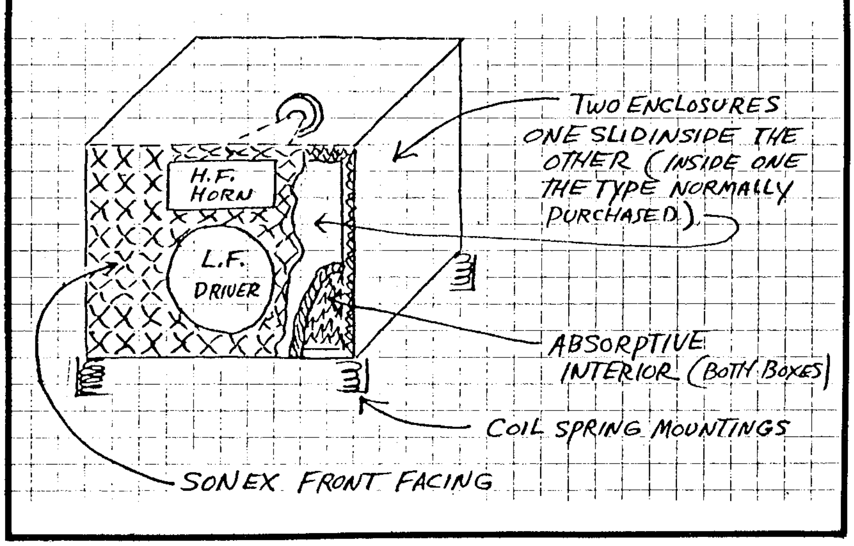
HOUSING LOUDSPEAKER SYSTEMS

Regenerative feedback in a sound system can have many origins -- proximity of circuits of differing levels, coupling through the air, microphonic components (i.e., turntables), etc.

Properly mounting and enclosing a loudspeaker system can help insure freedom from structure borne feedback sources and comb filters associated with the array itself. Comb filters originating at the array are detected by the ability to move a live microphone without a change in the feedback frequency. If a slight movement of the microphone causes a shift in the feedback frequency, then it usually is a comb filter associated with the microphone placement.

It has been demonstrated many times that high frequency drivers and their horns should have an enclosure around them. Spurious radiation from both driver and horn can, on occasion, be surprisingly high level and directive. Diffraction effects and inadvertent reflectors (grills, border trim, etc.) can create unexpected and, if not measured "in situ," unsuspected anomalies in the system's response. Therefore, it is wisdom to enclose all components. In the case of already housed units, their enclosure is slid into a second enclosure. The front of the main exterior housing should be totally absorptive with the loudspeakers brought as near flush to the surface as possible. Since reflections within 2 feet are the most dangerous (widest bandwidth anomalies are caused by short length reflections of equal level to the direct sound), every effort is made to have the absorption reach a minimum of 2 feet on every side of the loudspeakers.

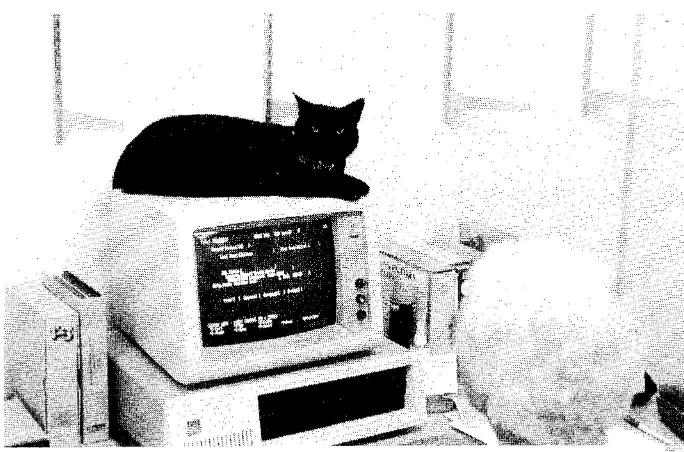
HOUSINGS FOR LOUDSPEAKER SYSTEMS

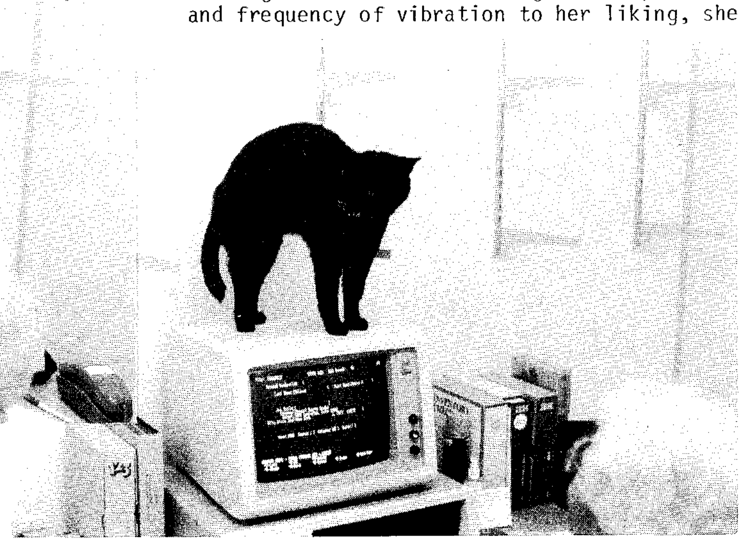


Mason Industries of Hollis, NY, and Los Angeles, CA., provides an extremely wide range of shock mountings for devices like loudspeakers along with excellent accurate individual application advice. These include shock mountings with suitable "snubbers" for earthquake prone areas where a heavy array could conceivably get into a non-linear swaying. When such a non-linearity occurs over a predetermined amplitude, the snubbers "lock up" the mountings. The construction of array mountings is an art and science of its own and we always recommend the services of an experienced professional registered engineer to approve final plans and construction. ♦

NEVER OPERATE WITHOUT A BACKUP

Francine, our black "office" cat, has taken a liking to the IBM-PC used in our office for word processing and mailing lists. After finding the temperature and frequency of vibration to her liking, she talked Carolyn into listing the kittens she has nursed. The second photo was her response to "You've lost what?" followed by "Of course, I don't have a backup." ♦





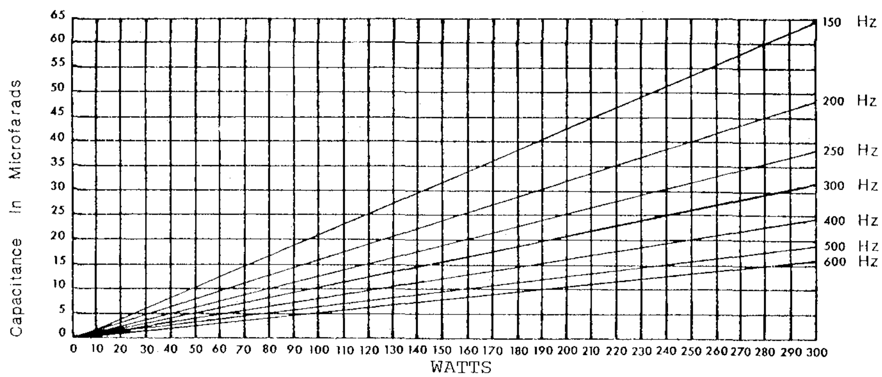
HOW TO DISTRIBUTE POWER

Rauland-Borg Corporation, Chicago, IL., recently sent the following very useful graphs to their distributors.

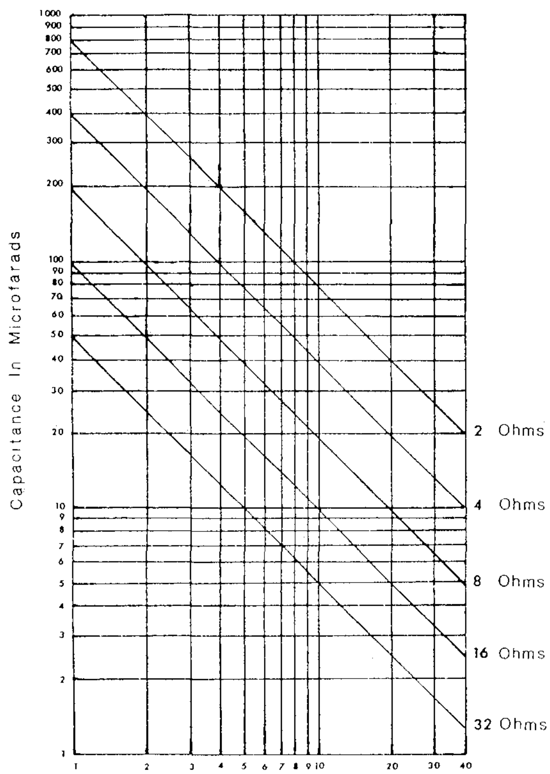
These duplicate equations in sound system engineering and free the user from the use of an electronic calculator. ♦

70.7V Speaker Line

On Protection Frequency Curve
At Speaker Wattage Read Capacitor Value



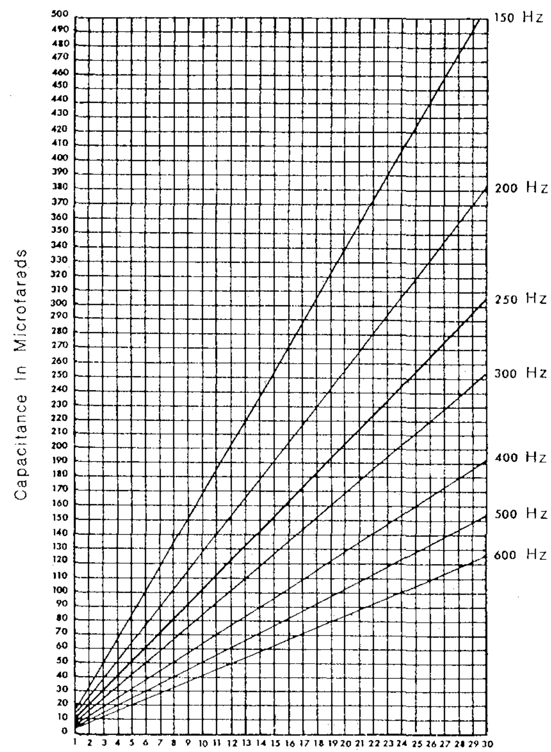
On Impedance Curve
At Protection Frequency
Read Capacitor Value



Frequency x 100

25V Speaker Line

On Protection Frequency Curve
At Speaker Wattage Read Capacitor Value



Watts x 10

A GENUINE "SIGN OF THE TIMES"

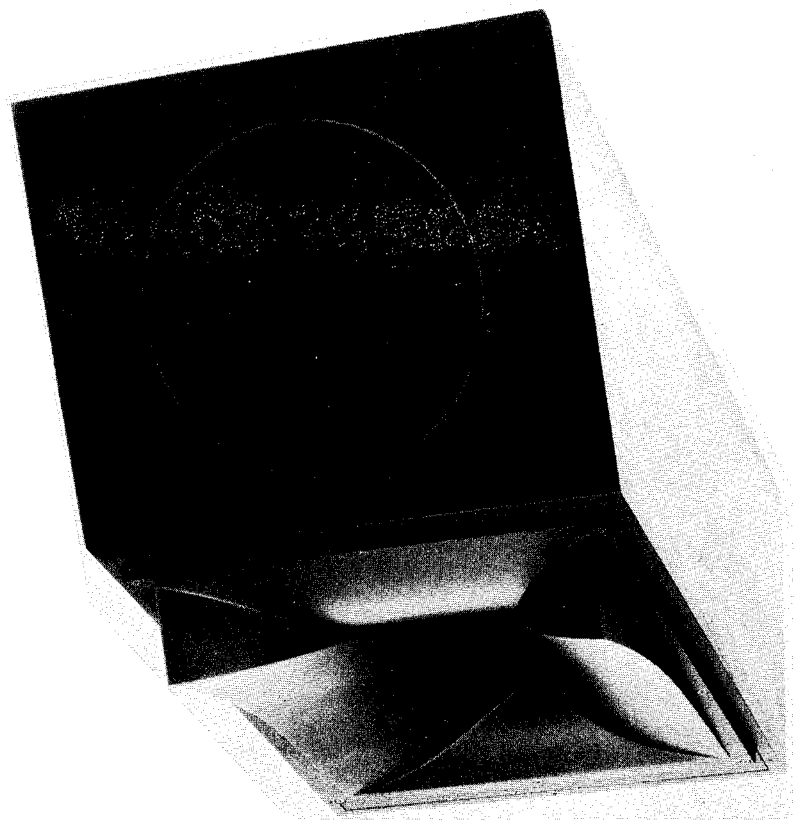
A radio advertisement offers "CLEO." The aerospace companies working on "Laser" weapons have a *free* bulletin board accessible to users of personal computers that lists jobs available to scientists, engineers and technicians. It is an interactive program allowing the individual to input his or her qualifications relative to job offerings and, in essence, "interview" the company.

This seems to us to be an ingenious way to prequalify applicants for work involving an understanding of programming. To all those who might feel that Syn-Aud-Con tends to over emphasize the importance of HP-41s and TEF® analyzers to an individual's audio future, this is certainly a significant "cloud on the horizon." ♦

JBL'S NEW 4660

For years, sound reinforcement system designers have wished for high-frequency horns whose patterns were specifically tailored to cover irregular or skewed areas. Until quite recently, all available hardware has exhibited pattern symmetry about both horizontal and vertical axes, and this meant that two or more devices had to be specified in order to get proper near and far coverage in the typical rectangular room.

At last fall's AES Convention, Don Keele of JBL presented a paper on a new horn design which gave remarkably even coverage over an area 2 units wide and 2.7 units long. The horn location was one unit high at one of the small ends of the room. Measurements showed that this horn's response around the periphery of the space was constant within a 4 dB window. This model demonstrated the capability of the design program, and Keele immediately set about to design a somewhat smaller device, with an 800 Hz crossover, that could become the HF section of a two-way system to be used in general sound reinforcement applications.



JBL - 4660
Defined Coverage Loudspeaker System

The new horn is now the HF section of the 4660 system. It was designed to cover, along the -6 dB isobar, a space 2 units wide and 2.6 units long. The horn location is assumed to be at a height of 0.6 units at the narrow end of the room, and moved back 0.35 units from the front edge of the desired coverage area. These conditions correspond quite closely to a typical house of worship, and allow for the system to be installed well toward the back wall, with coverage commencing at the front row in the seating area.

This particular horn has a coverage angle (-6 dB) of 110 degrees at the front, narrowing to 38 degrees at the back.

Examining the photo of the 4660, we note that the assembly is made to be mounted against a flat surface, the ceiling. The LF transducer in the system is the JBL 2225H, and it is angled downward toward the center of the target area. The HF driver is the JBL 2425J.

We are at the beginning of an interesting era with this new horn, and a careful assessment of the market for the 4660 will determine the future directions of these unique devices. In the meantime, nothing would surprise us in the way of creative usage of the 4660 in the hands of competent system designers. For example, what about locating a stereo pair of these systems on a wall, cross-firing them so that the stereo phantom center image remains fixed over a large listening area? (From Ken Lopez - JBL) ♦

PAUL KLIPSCH

Paul Klipsch has been inducted into the Audio Hall of Fame.

We are not sure what the Audio Hall of Fame is, but we sure know who Paul Klipsch is! That keen engineering mind hidden inside the masquerade called Paul Klipsch is indeed worthy of all and any accolades.

It is a constant delight to Syn-Aud-Con that the jealous and envious have been unable to topple his stature. It's reassuring to know that such individualism can still survive (Paul's 80 years old) and gains new vigor each year, apparently receiving only fertilization from the level of competitors in his business.♦

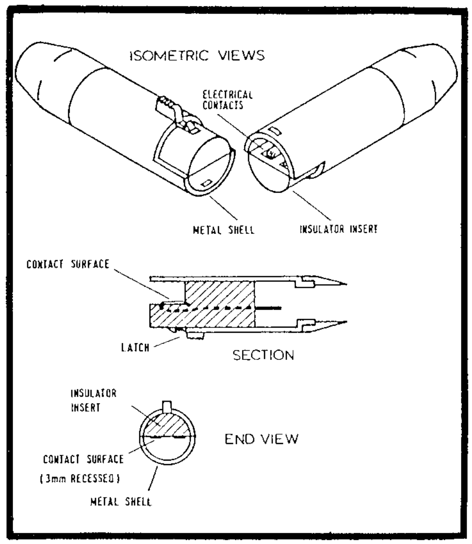
HOUSTON - NOV. 29-30, 1983

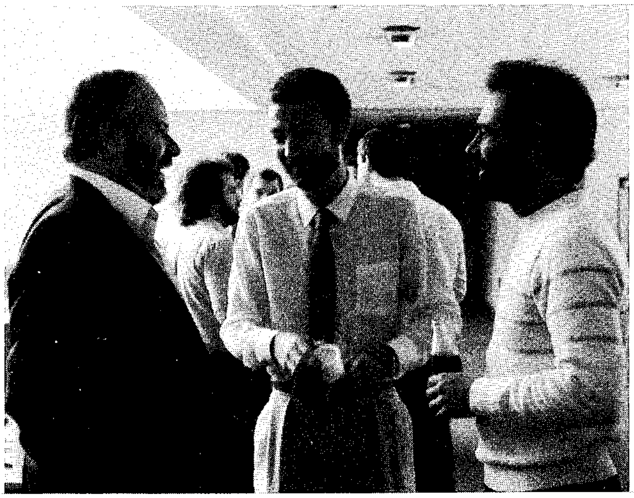
LAS VEGAS - DEC. 13-14, 1983

GENDERLESS XLR-TYPE CONNECTOR

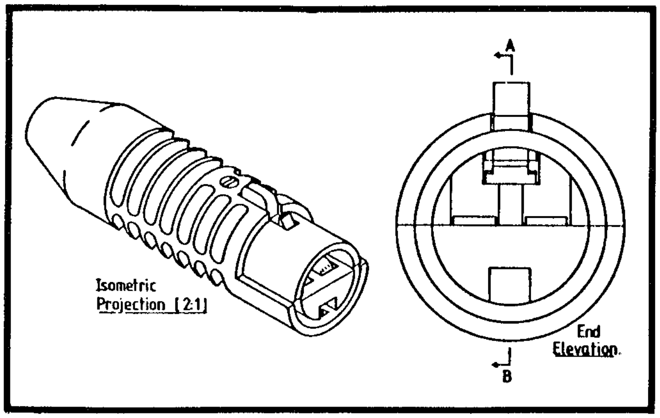
Syn-Aud-Con grads from Australia, Mike Dixon and Robert Grunberg, are patenting a new genderless XLR-type audio power connector (amplifier to loudspeaker). This means that a user would not have to stock male and female connectors. As an added benefit, it handles high current. The same connector can be used on ends of the cable. The connector seems to have everything to recommend it. The obvious negative factor is that there is enormous inertia to change within the cable and connector industry.

If anyone has "connections" within the industry or any ideas as to how to "sell" the industry on this new connector, contact Syn-Aud-Con. Robert Grunberg and Michael Dixon will be in the United States until July. After that, contact Dixon Design & Development, P. O. Box 462, Woollahra, NSW 2025 Australia. Tel: (02)389-5309. ♦





Mike Dixon (C) and Robert Grunberg (R) of Australia talking to Graham Thirkell of New Zealand during the recent TEF® Workshop.



DEPICTING THE LEVEL OF SOUND FIELDS

The sound field levels of interest to the sound system designer are:

1. The direct sound field level (L_D)
2. The early reflected sound field level (L_{RE})
3. The reverberant sound field level (L_R)
4. The ambient noise field level (L_N)

These sound fields can exhibit dependence upon distance, time and frequency. They can be depicted as:

1. Nature of sound field versus frequency
2. Nature of sound field versus time (ETC)
3. Level of sound versus distance (H.S.)
4. Level of sound versus frequency (NC curves)

The *total* sound field level (L_T) is:

$$L_T = 10 \text{ LOG } (\text{Exp } L_D/10 + \text{Exp } L_{RE}/10 + \text{Exp } L_R/10 + \text{Exp } L_N/10)$$

Notes

A. In summing levels on a power basis, it is not necessary to go to an absolute power ratio, but merely to a relative power ratio.

B. It is common practice to combine L_R and L_{RE} into a single value and label it L_R.

C. Some users consider L_D and L_{RE} out to 50 msec as L_D.

D. These levels may appear together, separately, or in various combinations. Part of the measurement process should be to identify which sound fields are present and what are their levels. ♦

COMMUNITY ON THE MOVE

Cluster Computer™

Community is shipping (and by now has filled all backorders) the materials for John Prohs' Spherical Loudspeaker Mapping Technique.

The basic Cluster Computer consists of:

1. A license for the use of the Cluster Computer™ process (patent pending).
2. Two transparent hemispheres, enscribed at 5^{0} intervals along the latitude and longitude.
3. Tape overlays that are placed outside the hemispheres to easily identify angle and elevation.
4. Program cards to upload an HP-41C, CV or CX with the programs: Sphere™, Point™ and Angle™.
5. Transparent isobar overlays of selected Community Light & Sound products.
6. Instructions and operating manual.

The cost for the license, hardware and software for the basic Cluster Computer is $195.00, plus $4.00 postage and handling. Additional hemispheres, overlays of different manufacturers' products, light sources and hard copy accessories will be available directly from Community Light & Sound. Prices for these additional products will be announced at a later date.

Architectural mapping is a term to describe any process that converts architectural room dimensions of an audience area in lineal feet or meters into an equivalent set of parameters describing the same areas as viewed from a chosen location for an electroacoustic array in angular (in degrees) range (in dB) notation on a suitable calibrated projection grid (usually one with latitude and longitude notation).

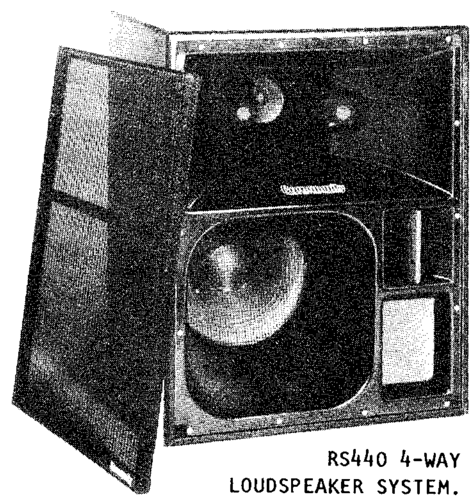
The Sphere, in addition to solving the severe distortions at high angles inherent in the "flat earth" techniques, allows not only the vertical, horizontal or diagonal angles to be accurately obtained, but allows the rotational angles as well (not possible in the other techniques).

This is a very powerful tool. Our industry is fortunate to have a tool available to us at such an extremely low cost. If you have been reading news releases in the audio press, you know that acoustical consultant, David Klepper, who got a prototype sphere from John Prohs, has designed many successful cluster designs with the new program.

320 Processor

The following was written by Bruce Howze of Community:

Community Light & Sound has developed an interesting signal processor for use with its RS320, a compact loudspeaker system. The RS320 was intended primarily for sound reinforcement applications - the major design goals being sensitivity, intelligibility and projection. With a mid-band sensitivity of 104dB 1W/1M resulting from its 2" throat midrange horn and driver, the system succeeds in achieving these goals, but its response is not exactly flat. The HF and LF sections are simply not equal to the midrange capability. It was decided to provide the system with a line-level equalizer to solve the response problems rather than to pad down the midrange in the crossover. This choice was made to avoid the considerable waste of amplifier power that would result from such a pad. Production RS320's are equipped with a switchable midrange pad, so that operation with equalization is not mandatory.

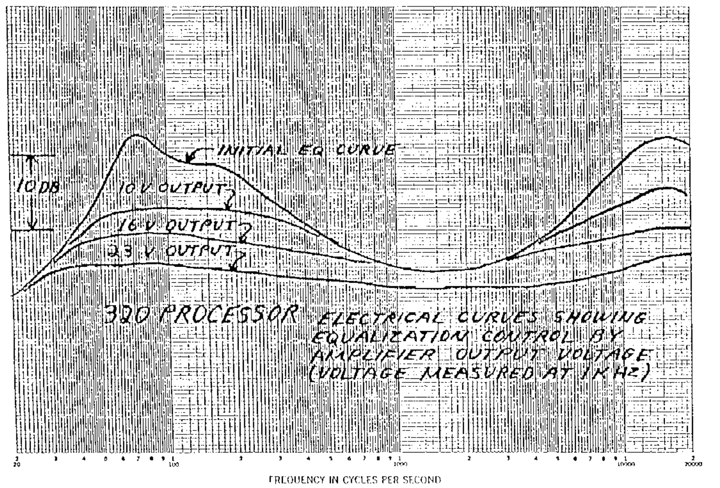


RS440 4-WAY LOUDSPEAKER SYSTEM.

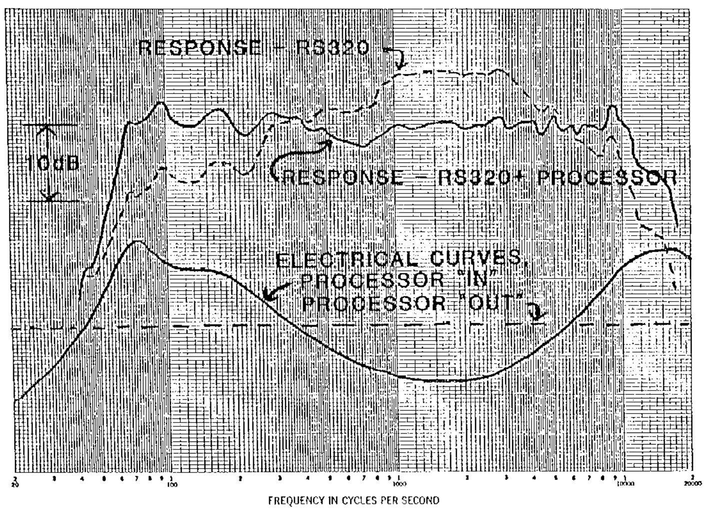
The use of front-end equalizers to help out the response of compact speakers is certainly not a new idea; Bose popularized the concept quite a while ago. The approach works well in its objective of providing flat response but can cause major difficulties in the headroom department: with the 12 to 16dB of boost needed at the low and high frequency ends of the spectrum, the power amplifier will go into clipping at even modest acoustic output levels, which severely limits the system's utility for high level applications. With a fixed equalization unit, only two solutions to this problem seem possible: either employ a gigantic amplifier or use less boost and accept the resulting response deficiencies. In the 320 Processor, Community tried a third solution - the equalizer monitors the power amplifier output and continuously adjusts its equalization curve, keeping the amplifier fairly well out of clipping.

The thresholds and rates of de-equalization were empirically set by extensive listening tests to achieve a good balance between system response and amplifier distortion. The performance of this dynamic control approach is a significant improvement over a fixed EQ system - the acoustical output is remarkably flat at moderate levels (112dB, 1M), and at higher levels the gradual low and high frequency roll-off enables a steady output of 125dB with peaks well over 130dB - not bad for a moderately priced loudspeaker that weighs less than 50 pounds.

Continued next page.....



Measured Object RS320 + 320 PROCESSOR Test Procedure
Measurement Scale 10 dB/IN. Date 11/6/83 Initials BRUCE
Notes: AKG MIC AT .18", 30 SEC SINE SWEEP



Control circuitry in the 320 Processor separately monitors the low, mid and HF output of the power amplifier and controls the appropriate area of the equalization curve. The accompanying electrical curves show the initial equalization and gradual flattening thereof as output level increases. Once a level is reached such that the curve is essentially flat, an overall compression mode is initiated and further level increase simply lowers the system gain. Front panel controls provide selection of de-equalization threshold voltages to match various amplifier output capabilities and also provide a choice of four different initial equalization curves. Although the 320 Processor was specifically designed for use with the RS320 loudspeaker, it could also be used to advantage with other systems requiring similar equalization.

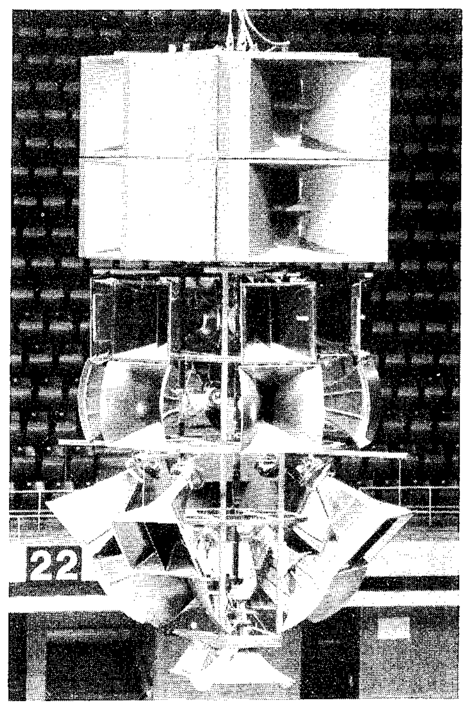
TEF® Analyzer at Community

Bruce Howze gave us permission to tell of their first use of the TEF® analyzer. We were holding a Loudspeaker Array Workshop in April 1983 and Bruce was scheduled to attend. Knowing that their first Pattern Control horns were in prototype, we invited Community to bring along one to demonstrate and measure during the class. A few weeks before the class we received a phone call from Bruce saying that the TEF analyzer revealed problems in the new horns (problems, by the way, that exist in most "constant directivity" type horns) that weren't evident in conventional measurements. As Bruce said, "It's one thing to release a product not knowing a problem exists; it's another to release it knowing it's there." So it was back to the old drawing board.

Bruce wrote:

The design of the Community PC horns relied heavily on data from a Crown TEF® machine. Waterfall dispersion plots provided a picture of horn performance in such detail that the slightest aberration could be seen and investigated. With this analytical tool, the initial horn designs were optimized by a combination of mathematical and empirical methods to meet the assigned performance goals while keeping the horn size at a minimum for ease and convenience of clustering. Community has used these same methods to develop three 2" throat pattern control horns for Rauland-Borg.

The TEF analyzer was also used to investigate the beaming tendencies of bass horns. At the upper end of their ranges typical bass horns have very narrow patterns. This can cause major directivity mismatches, particularly in two-way systems with crossover in the 500-800 Hz range. Community has developed two new LF units designed to provide uniform coverage in the crossover region: The CB594 is a 50 Hz all fiberglas straight bass horn for a single 15" or 18" driver. The VB664 is a compact vented horn employing dual 15" drivers. Both horns have $60^{\circ} \times 40^{\circ}$ patterns in the 500-800 Hz range.



Jake Ewalt, Technical Director of Sound Engineering at Iowa State University, designed the cluster shown above and installed at the J. H. Hilton Colosseum. The cluster uses Community horns and drivers (including the M4 driver) and JBL bass horns (from an earlier installation). Jake says some compromises were made to make the cluster movable.

Literature

Look at the new literature from Community. Their literature contains far more information than typical literature. All that is changing. TEF® measurements are creeping into loudspeaker literature. Emilar used waterfalls in their new literature. Electro-Voice has a TEF analyzer as does Shure Brothers. We want to see linear frequency curves (they can run both log and linear for comparison) because it is linear measurements that tell us what is truly going on. Community is on the move and is leading the way. ♦

LOUDSPEAKER IMPEDANCE MEASUREMENTS

Tecron demonstrated its special impedance measuring cable (measures accurately up to 200Ω) (see Fig. #1) for the TEF® analyzer. The uniqueness of this cable is that:

1. The loudspeaker is driven directly from the TEF analyzer.

2. 2 volts open circuit output results in a reading of 1Ω per millivolt, thus allowing the impedance magnitude, phase, and frequency plus the reactive and resistive values to be read directly from the cursor's screen notation.

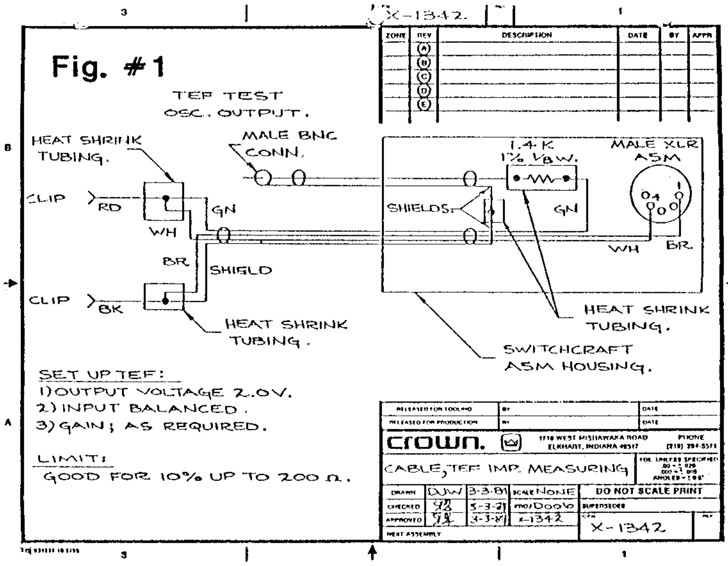


Fig. #1

Some Example Measurements*

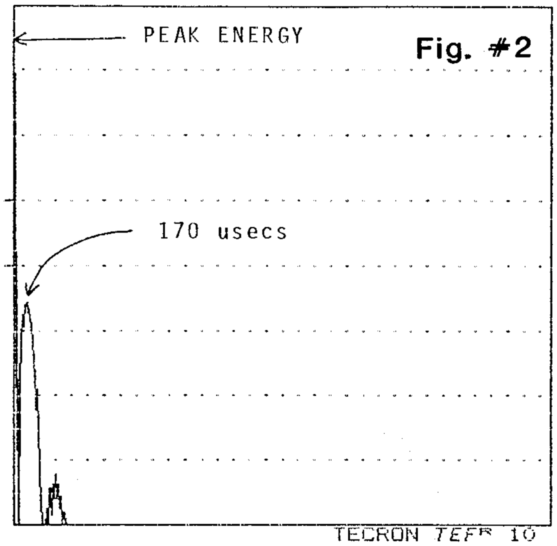


Fig. #2

Fig. #2

Vertical: 6dB/div with base of display at 30.8dB
0dB is located at .00002 Pascals

Horizontal: 0 microseconds or 0 FT to
6278 microseconds or 7.09364 FT
scale: 1.9395E+00 FT/inch or 7.6357E-01 FT/cm.
1716 microseconds/inch or 675 micro-seconds/cm.

Line Spacing: 15.7332 microseconds or 1.77785E-2 FT
Line Width: 21.3971 microseconds or 2.41787E-2 FT

Sweep rate: 5009.55Hz/Sec

Sweep range: -31778.90Hz to 31781.00Hz

Fig. #2 is the energy time curve (ETC) of the impedance of a small Radio Shack loudspeaker. It's evident that sufficiently short transients don't see an impedance but pass straight through the system.

Fig. #3

Resolution: 7.9720E+01 FT & 1.4175E+01Hz

12dB of automatic screen gain.

Frequency range: 69.60Hz to 10001.20Hz

Time of test: 0 microseconds, 0.0000E+00 FT

Sweep Rate & Bandwidth: 200.92Hz/Sec & 1.4175E+01Hz

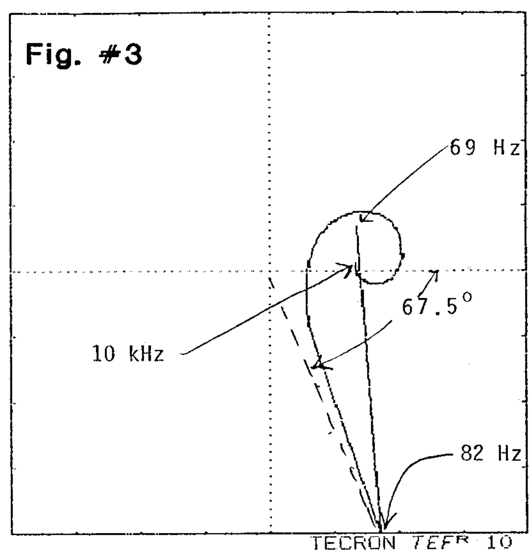


Fig. #3

The Nyquist phase plot (NPP) (see Fig. #3) of the resistive and reactive components of the impedance shows a low frequency capacitive reactance phase angle of approximately -67.5°. Note also that we do not have sufficient resolution. The sweeping rate is too fast at low frequencies because we want a quick overview of which frequency ranges to sweep slowly later.

Fig. #4 is the "magnitude" of the impedance from 70 to 10,000 Hz. Fig. #5 is the phase angle vs frequency curve (PFC). At this point in the measurement process we chose to sweep from 70 Hz to 248 Hz at a sweep rate of 5 Hz/sec with a bandwidth of 2.2 Hz. If we had swept

*All Sample Measurements are Syn-Aud-Con tests done by Don Davis on 4/8/84 at Rancho Carrillo, California.

Continued next page......

the wider range at this rate, each sweep would have taken: $\frac{(10{,}000 - 70)}{5 \cdot 60} = 33.1$ minutes

As it was, the sweep took: $\frac{248 - 70}{5} = 35.6$ seconds

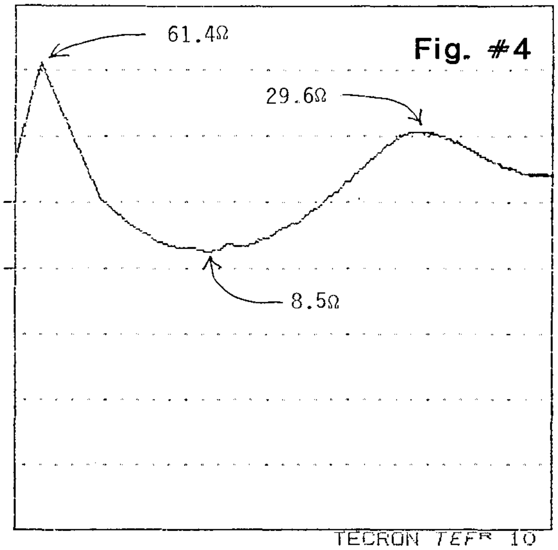


Fig. #4

Vertical: 6dB/div with base of display at 30.9dB
OdB is located at .00002 Pascals

Horizontal: 69.60Hz to 10001.20Hz
Log freq axis (2.7decades)

Resolution: 7.9720E+01 FT & 1.4175E+01Hz

Time of test: 0 microseconds, 0.0000E+00 FT

Sweep Rate & Bandwidth: 200.92Hz/Sec & 1.4175E+01Hz

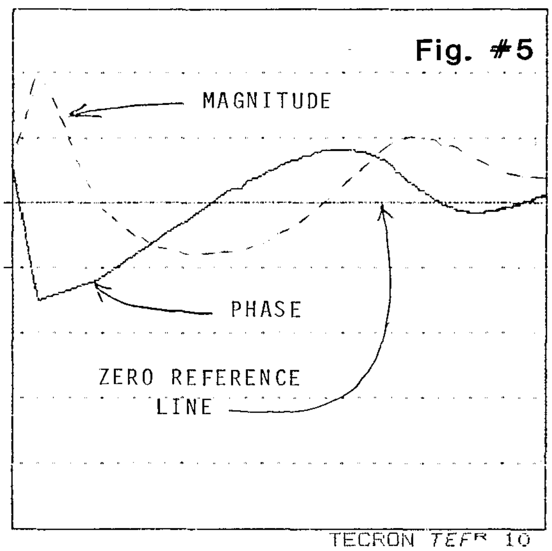


Fig. #5

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

Horizontal: 69.60Hz to 10001.20Hz
Log freq axis (2.7decades)

Resolution: 7.9720E+01 FT & 1.4175E+01Hz

Time of test: 0 microseconds, 0.0000E+00 FT

Sweep Rate & Bandwidth: 200.92Hz/Sec & 1.4175E+01Hz

Note here that we now have a much higher resolution measurement of both the frequency and the magnitude of the resonant peak. (See Fig. #6) It takes 84 Hz 12 msecs to produce a wavelength and our T_R for this measurement was:

$$\frac{2.2 \text{ Hz}}{5 \text{ Hz/sec}} = 440 \text{ msec.}$$

Our earlier measurement had a:

$$T_R = \frac{14 \text{ Hz}}{200 \text{ Hz/sec}} = 70 \text{ msec.}$$

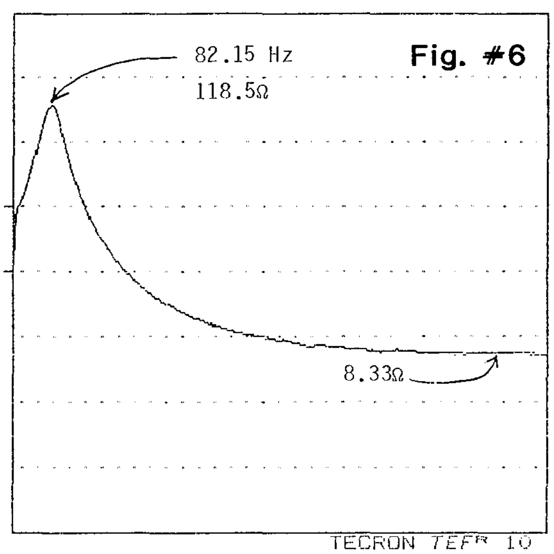


Fig. #6

Vertical: 6dB/div with base of display at 39.9dB
OdB is located at .00002 Pascals

Horizontal: 69.60Hz to 248.42Hz
scale: 48.89Hz/inch or 19.25Hz/cm.

Resolution: 5.0487E+02 FT & 2.2382E+00Hz

Time of test: 0 microseconds, 0.0000E+00 FT

Sweep Rate & Bandwidth: 5.01Hz/Sec & 2.2382E+00Hz

Fig. #7 is the high frequency resonant peak done at a slower sweep and higher frequency resolution. Note here that the high frequency resolution was sufficient in the earlier measurement. These last two figures both have *linear* frequency scales whereas the overview figures are logarithmic frequency scales.

Continued next page.....

Fig. #7

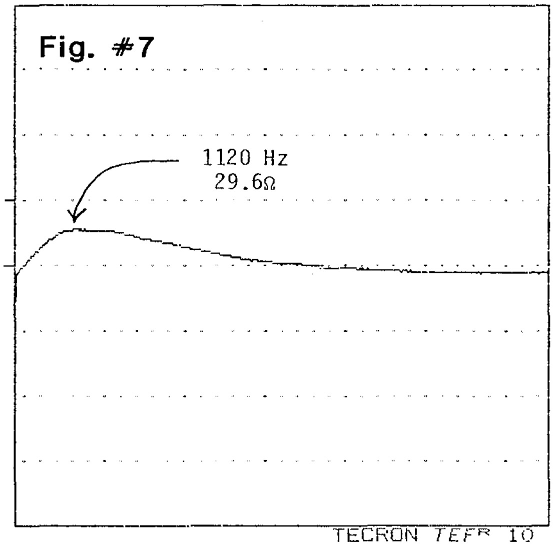
Vertical: 6dB/div with base of display at 39.9dB
0dB is located at .00002 Pascals

Horizontal: 1000.12Hz to 10001.20Hz
scale: 2460.98Hz/inch or 968.89Hz/cm.

Resolution: 5.0487E+01 FT & 2.2382E+01Hz

Time of test: 0 microseconds, 0.0000E+00 FT

Sweep Rate & Bandwidth: 50.10Hz/Sec & 2.2382E+00Hz



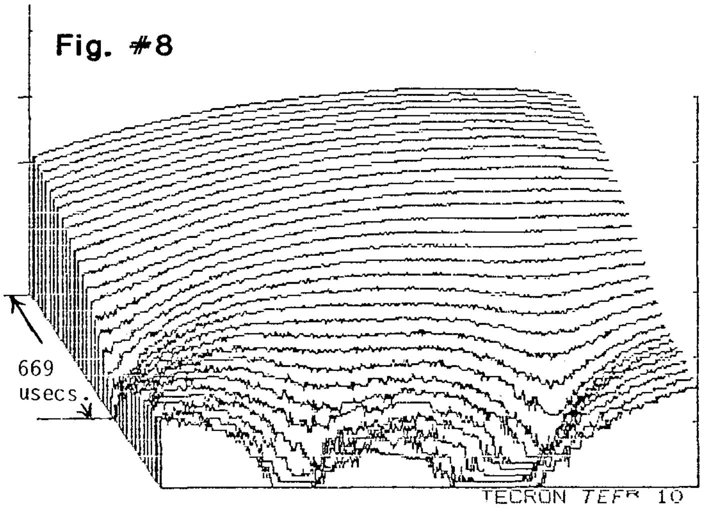


Fig. #8

Vertical:
12dB/div with base of display at 27.9dB
0dB is located at .00002 Pascals

Horizontal: 69.60Hz to 2501.37Hz
scale: 664.87Hz/inch or 261.76Hz/cm.

Resolution: 5.5640E-01 FT & 2.0309E+03Hz

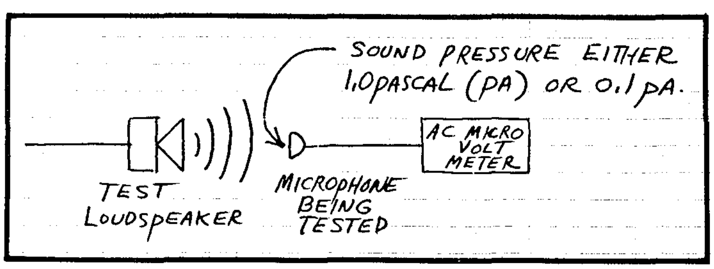
Time of test:
1101 microseconds 1.2445E+00 FT (front)
to 0 microseconds 0.0000E+00 FT (back)
-36 microseconds/step or -.0401449907069 FT

Sweep Rate & Bandwidth:
2030.90Hz/Sec & 1.0000E+00Hz
(T_ω = 492 usecs)

Finally, a 3-D view of the impedance magnitude. (See Fig. #8) An interesting point here is that the time window T_W being used here is 493 usecs. This T_W is being "stepped along" at 36 usecs per step. Thus, the low frequency dip at the low end is 669 usecs. minus 493 usecs. equals 176 usecs., i.e., the value shown on the ETC.

Generation of a set of meaningful reference curves for loudspeaker designers could be one of those projects worth a TEF analyzer user's spare time. ♦

HOW MICROPHONE SENSITIVITY IS MEASURED



1. Test loudspeaker generates test sound pressure *at the diaphragm* of the microphone being tested.

2. The open circuit voltage from the microphone is read on the AC microvolt meter.

NOTES

$$\text{Microphone sensitivity} = \frac{\text{Open Circuit Voltage}}{\text{Test Sound Pressure}}$$

1.0 pascal (Pa) = 10 dynes/cm^2 = 10 μ bar = 10 newton/m^2

0.1 pascal (Pa) = 1.0 dyne/cm^2 = 1.0 μ bar = 1.0 newton/m^2

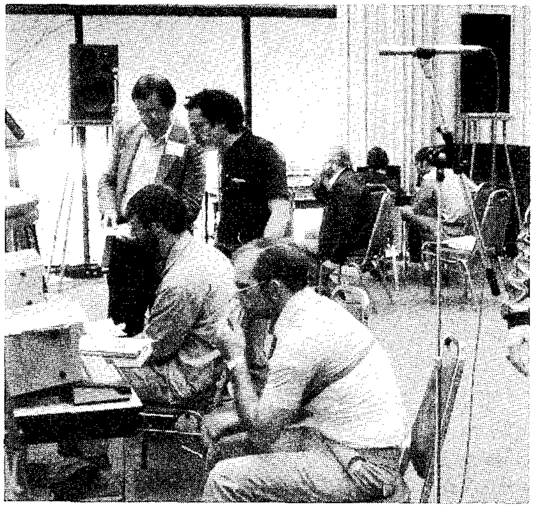
$$\text{Test } L_p = 20 \text{ Log}\left(\frac{1.0 \text{ Pa}}{0.00002 \text{ Pa}}\right) = 94 \text{ dB}$$

$$\text{or: } 20 \text{ Log}\left(\frac{0.1 \text{ Pa}}{0.00002 \text{ Pa}}\right) = 74 \text{ dB}$$

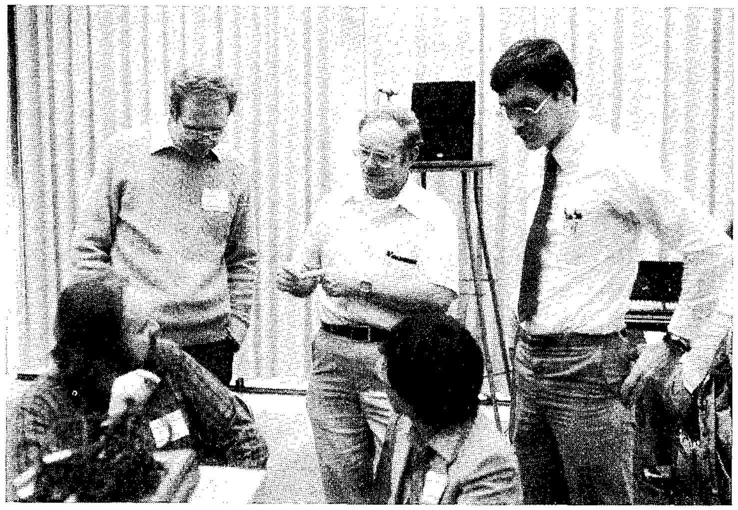
♦

TURNTABLE FROM J. W. DAVIS & COMPANY

We were explaining in the presence of Harvey Earp, President of J. W. Davis & Company, how important it is to make loudspeaker measurements without nearby reflecting surfaces interfering with the measurements, especially low frequency measurements. Harvey said, "What you need is a turntable to hold the loudspeakers." A few weeks went by and Harvey called to say, "I've got something for you," and thus was born an inexpensive, extremely valuable tool for anyone making loudspeaker/acoustic measurements.



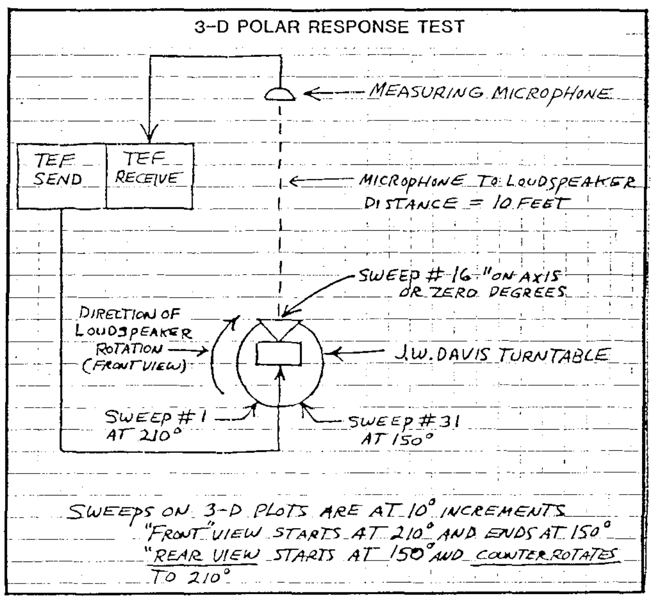
Turntables in use during the March TEF class.

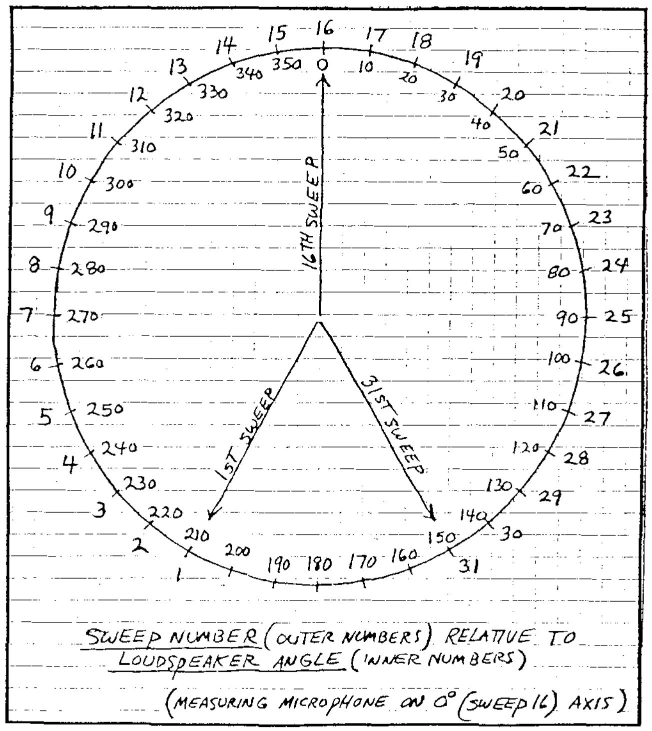


A Spica loudspeaker on the turntable for measurement.

The turntable allows 31 sweeps in 10 degree increments for polar plots. We ordered five of the Davis turntables for our March TEF® Workshop, which were made up for Syn-Aud-Con, with the agreement that we could sell them at a reduced price (normally $250 plus shipping) after the workshop. Our price is $225 including shipping. ♦

The two figures reproduced below are from Newsletter Volume 10, Number 3/4, page 18.





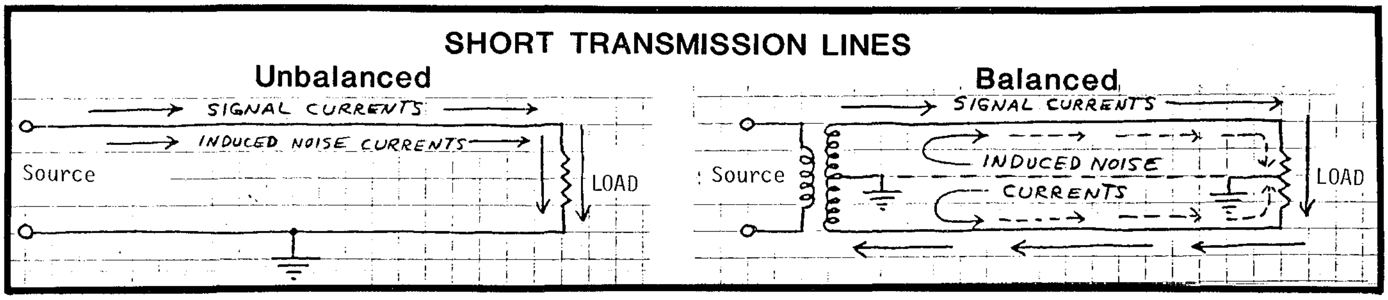
READY! FIRE! AIM!

Electronics/April 5, 1984, "Marketing in High-Tech Start-Ups." A marketing consultant wrote about many new businesses in high-tech failing because of the failure to properly market the product.

He made the analogy: Building the product ("Ready!") and then "selling the hell out of it." ("Fire!") Marketing is the aiming--the "aiming" that is done before the "Fire!" (selling). ♦

BALANCED AND UNBALANCED LINES

An *unbalanced line* is a line in which one of the two conductors is at or near ground or zero potential, while the other, known as the "hot" or "high" side of the line, is at a higher potential. Since both conductors are at unequal potentials with respect to ground, the line is said to be unbalanced. Low cost, high impedance microphone circuits consisting of a single conductor and shield are an example of an unbalanced circuit.



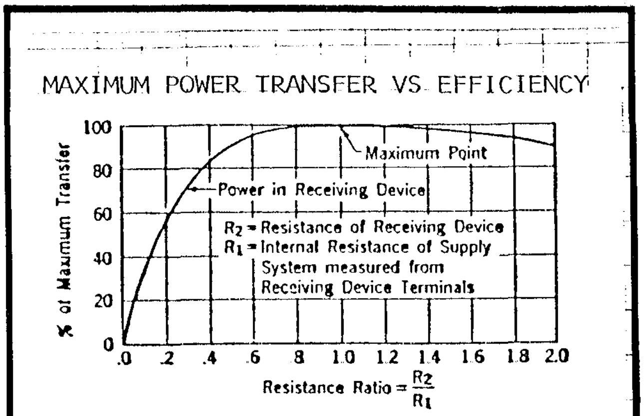
A *balanced line* consists of two identical conductors that have the same voltage, except that the polarity is opposite with respect to ground. Balanced circuits require three terminals rather than two. The center terminal is considered to be neutral and is usually connected to ground. The two outer conductors (on either side of the center) are at high signal potentials with respect to ground. Note particularly that their potentials are the same and are of opposite polarity. Most balanced lines are driven from a center-tapped transformer.

At the receiving end the load is divided into two parts with a ground connection between them. On some balanced lines there is no center tap, but as long as the line has equal potentials on its two conductors, it is considered to be balanced. 70 volt loudspeaker lines are a good example of a two conductor balanced line.

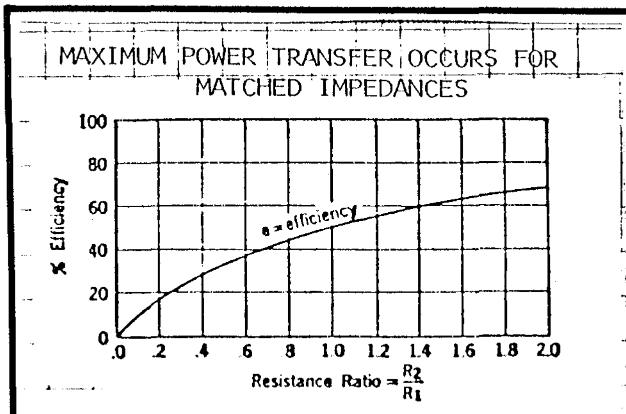
Balanced lines are very useful in eliminating or reducing unwanted induced noises. Induced signals having the same polarity in both lines (the noise currents *flow in the same direction* in both lines) are, at the load, found to be in opposition and of equal amplitude so they cancel completely.

The Pros And Cons Of Matched And Unmatched

Matched circuits provide maximum power transfer. An unmatched circuit where an R_S is substantially lower than an R_L yields a higher transfer efficiency, i.e., causes less of a load on the source while obtaining the desired *voltage* amplitude, and a better noise figure by 3.01 dB. Where reactive circuits are involved, such as passive equalizers, conventional crossover networks, and occasional active circuits as well, where isolation stages are not provided at the input and output, critical matching of input and output impedances becomes necessary.



POWER TRANSFER AS A FUNCTION OF RESISTANCE RATIO.



EFFICIENCY AS A FUNCTION OF RESISTANCE RATIO.

Greatest efficiency occurs for $R_L > R_S$ (Note efficiency difficulties for the case of $R_S > R_L$).

Definition Of Matched Impedances

Impedance matching is defined as: "The connection across a source impedance of a matching impedance that allows optimum undistorted energy transfer." Types of matching are:

1. An impedance having the same magnitude and phase angle as the source impedance. (In most conventional passive audio devices, the interface is essentially resistive and the phase angle low.)

Continued next page...

2. An impedance having the same magnitude and having a phase angle with the same magnitude, but opposite signs as the source impedance. This "conjugate" impedance corrects for the power factor (P.F.) caused by the reactance on the line where it has caused a detrimental phase angle relationship. This is, of course, normally a highly frequency dependent function.

Most "matched" impedances used in *audio systems* are for the magnitude of the impedance only. In *component engineering* careful accounting of the phase relationships must be maintained as well.

Appropriate Impedance

Rather than a "matched impedance", Z_{MATCH}, we can refer to the appropriate impedance, Z_a, and thus avoid contributing to the degradation of the term "matched" impedance where what is actually intended is the preferred load, R_L, or appropriate match for a given R_S. ♦

STATISTICAL VS SPECIFIC ABSORPTION

We know that in the case of small dead rooms (Recording Studios and control rooms) and in the case of extremely large rooms (the Super Dome, etc.) the statistical reverberation equations should not be utilized in any form. This does not mean that we don't require absorptive material, but merely that the amount and placement are not determined statistically.

In such spaces we can use the energy time curve ETC mode of the TEF® analyzer to find the vital data of:

1. direction of arrival
2. inverse square law level change along the reflected path (by comparison between regular ETC and 1/R ETC)
3. actual level from reflection at observer's position.

These are distinctly different requirements from those for rooms where the statistical reverberation equations are applicable and the parameters of

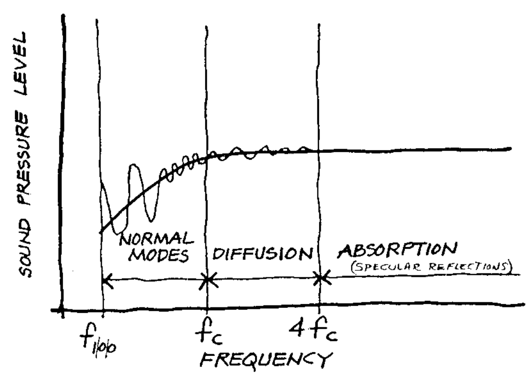
1. Volume V
2. Surface area S
3. Average absorption coefficient $\bar{a}$

are the tools used.

The case at hand can directly affect the use of absorptive material. For example, illustration (A) works well in statistical spaces, but (B) is found to work even better at low frequencies.

In small rooms, however, perhaps at the specular frequencies involved, (A) might be best because of the scattering it provides in addition to absorption.

CONTROLLERS OF STEADY-STATE ROOM ACOUSTIC RESPONSE

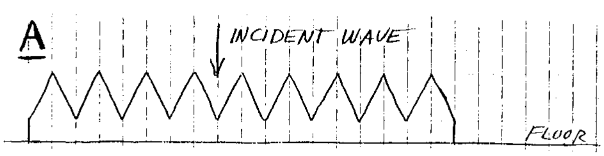


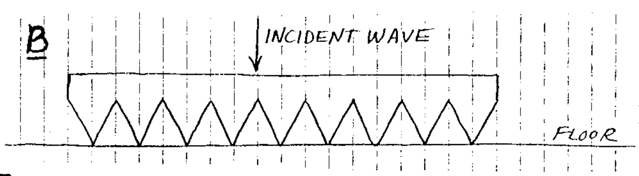
$$f_c = 11{,}250\sqrt{\frac{T}{V}}$$

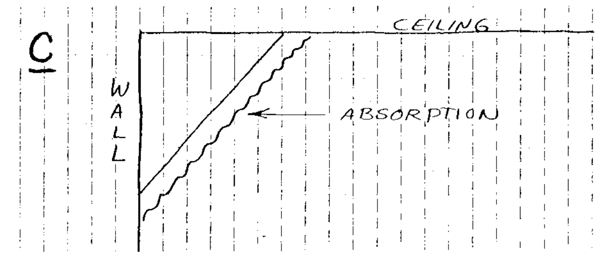
$$T = 0.049\frac{V}{S\bar{a}_{SAB}}$$

Courtesy of Bolt Beranek Newman, Inc.

NEW IDEAS ABOUT APPLYING ABSORPTIVE MATERIALS







Does the intersection placement shown in (C) work best for statistical reasons or because of the specific focusing of such intersecting surfaces?

Always have in mind when using absorptive materials whether you are dealing with room mode frequencies, diffusion frequencies or specular reflection frequencies. Then, consider if you are in a statistical case or a specific reflection case at the chosen frequency range and proceed accordingly. ♦

TELEPHONE INDUSTRY IN TROUBLE

An article in the April 1984 issue of the *IEEE "The Institute"* has a very dark view for the future of the telephone industry. At the IEEE Centennial Media Briefing, Dr. Howard Frank stated:

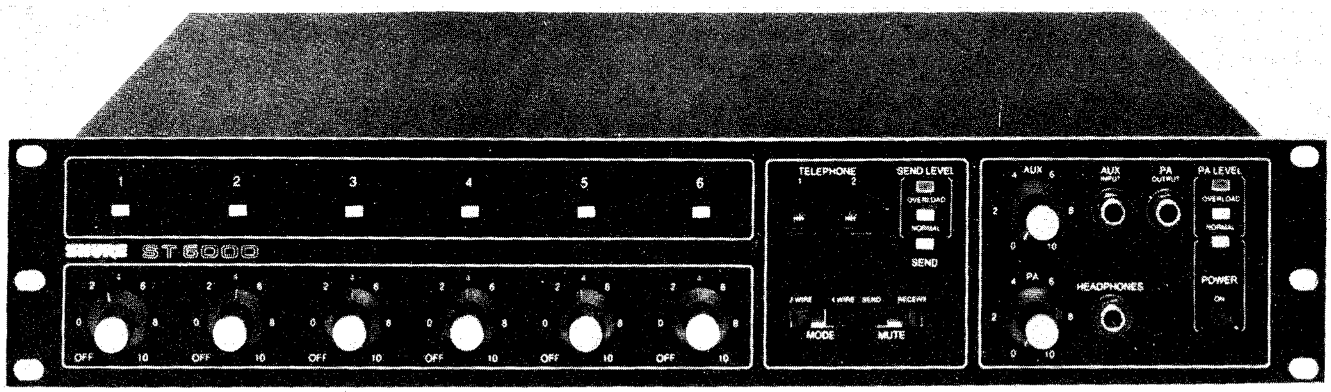
> *Incompatibility is now a key problem, he noted......The telephone industry, he said, has "degenerated into the condition the computer field was in 10 years ago, where every new computer introduced was incompatible with every other computer. The computer industry is still trying to climb out of that, and it will take another decade. The telephone industry is just stepping into that hole."*
>
> *Another major problem faced by the telephone industry today is personnel, Dr. Frank said. "The university system in the U.S. has failed to understand that the telecommunications industry in the 1980s is different from what it was in the 1960s," and as a result job openings for people with a telecommunications background remain unfilled.*
>
> *"And the problem is getting worse instead of better," he said. "The university system is silent in the telecommunications area."*

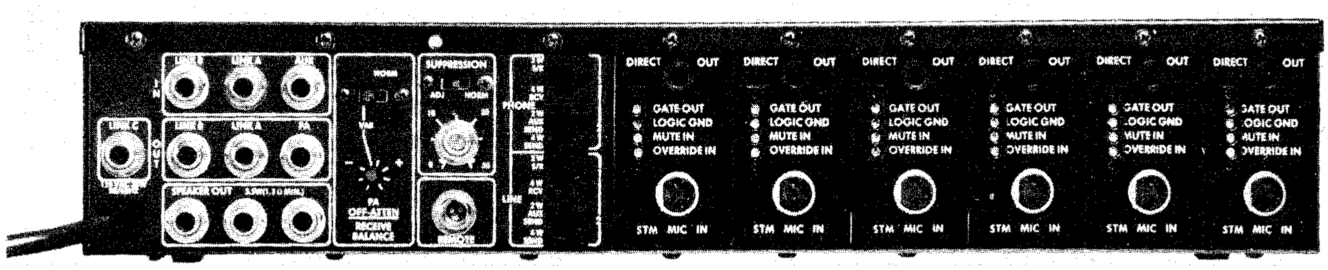
Where is this needed education and training to come from? Peter Drucker, in *U. S. News & World Report*, concluded the interview:

> Q: *Would you say that our schools are producing such technologists in enough numbers?*
>
> A: *No--though so far, whenever we've needed them, they have appeared. Partly it is because they fit into the genius of the American people and partly it is because, thank God, we have no ministry of education. Any fly-by-night outfit can start a course, and five years later the state colleges take it over, and 20 years later Harvard has a department with a Ph.D. program--by which time it's obsolete.*
>
> *So these people are appearing. Some learn their skills in the community colleges, but far more do so in free enterprise--places outside the official and approved and anointed educational system.* ♦

SHURE TELECONFERENCE SYSTEM

Alert innovative manufacturers are going to help "close the gap" by supplying good equipment. Shure Brothers, who are famous for their secrecy in research and development, has announced a fully developed line of telecommunications which they call "The Corporate Link™" (ST6000 System).





The Shure Teleconference System is a total system that consists of a number of complementary components, including a six-input console (ST6000), microphones (STM22, STM26, and STM28), loudspeakers (STL72 and STL78), a remote control unit, a video switcher interface (STV680), and an eight-channel expansion mixer (ST6008).

Shure is sufficiently serious about the future of teleconferencing needs that a new division of Shure has been formed.

Continued next page.....

The products include the latest technology, STM22 low-profile microphones with 120^{o} angle of acceptance, automatic mixer, the STL72 conference loudspeaker, a low-profile table top *downward-firing speaker* (italics mine). Shure literature says "the system also effectively *suppresses echoes encountered in satellite links* (italics mine).

If you haven't received literature from Shure for "The Corporate Link™," send for it. Shure Brothers is one of the largest professional sound manufacturers in the United States, one of the oldest, and personally directed by Mr. S. N. Shure, Chairman of the Board, and President Jim Kogen. It isn't uncommon to see an old company become stolid and peak in its growth, but Shure Brothers isn't a "common" company. ♦

NEW RULES FOR CONTRACTORS?

Sound & Video Contractor (January 1984) published an article by David L. Adams, consultant in acoustics in Denver, titled "New Rules for contractors? Division 17." Mr. Adams feels that it is time to take the sound and video specifications out of Division 16 (bid by electrical contractors). Mr. Adams says (excerpts):

> *Under the present situation, however, the cost of the sound and video system to the owner has been increased twice--once by the electrical contractor, who marks up the sound and video contractors' bid, and then again by the general contractor for the project. The designer of the system is now three times removed from the actual system installer, which makes coordination during and following construction very difficult.*
>
> *Because of their lack of knowledge of sound and video systems, most general and electrical contractors maintain an aloof, apathetic attitude regarding these "non-electrical" systems. For the designer and the sound and video contractor who must work through the architect, the general contractor and the electrical contractor, coordination of the system installation is far more complicated than it needs to be. With so many middlemen, the potential for error or misunderstanding is greatly increased.*
>
> *I urge everyone involved in the construction industry to make a concerted effort, through the national and local chapters of their respective trade and professional associations, to create a new Division 17 in the construction specification format. For you sound and video contractors, may I suggest you join and actively participate in the Construction Specification Institute. The consultants and the manufacturers of sound and video equipment are pulling for you.*

Vic Hall of Communications Company, Inc., in San Diego sent us a copy of his letter to *Sound & Video:*

> *Mr. Frederick J. Ampel* *Subject: New Rules for Contractor?* *March 2, 1984*
> *Sound & Video Contractor*
>
> *I wish to thank you for printing Mr. David Adams' article. As this industry has grown, more competent electronic, electro acoustic, and human engineering specialists are being trained. It would have been difficult or impossible only a few years ago to have found qualified independent engineers to prepare a Section 17 specification; not so today.*
>
> *Many of the abuses in our industry would diminish if our bids were made directly to the generals, not that their morality is any better than the electricians', but there are usually fewer of them. We Sound & Electronics Contractors regularly hear questions or statements like, "So and so bid ? , can you beat it?"; "Help me with a few thousand and we'll make it up next time."; or "Don't worry, we can get something cheaper approved."*
>
> *Qualified engineers could make these decisions and eliminate the temptation to cheat on the intent of poorly conceived and written specifications.*
>
> *The creation and use of Section 17 would eventually eliminate the unqualified engineer writing basically hardware specification that many times won't even work.*
>
> *I firmly believe that another of the problems created by otherwise knowledgeable consultants might even go away and that is, through their personal ego, designing such complex systems the customer and end user neither wants nor can use.*
>
> *I would like to suggest that sound contractors work through their associations, NSCA, IBMA, etc., to gain strength and unity for adoption of a policy advocating use of a Section 17 to specify all sound, electro acoustics, and video systems. This might include telephone, life safety, and security systems as well, or add the necessary sections for the various specialists required by both the engineering and contracting communities.*
>
> *Sincerely, Victor M. Hall*

Extremely knowledgeable and experienced Vic Hall has made several excellent points in his letter. ♦

IMPORTANT PAPER ON FEEDBACK

We are reproducing an excerpt from the 1969 paper by J. Ernest Benson and Donald F. Craig entitled *"A Feedback-Mode Analyzer/Suppressor Unit for Auditorium Sound System Stabilization"* that appeared in the March 1969 issue of the proceedings of the *IREE*, Australia.

This excerpt reveals a fundamental time domain behavior first predicted by Antman (whom they credit) and confirmed by them. The reason we are printing this excerpt is to insure that credit falls where credit is due, as last Fall's AES Convention had an attempted rediscovery of this data minus credits to earlier workers.

The obvious importance of this fundamental behavior is that we have a unique "tag" on the signal after it has been through the system and attempts to reenter the system. Enough said!

We highly recommend anyone interested in the equalization of sound systems to read this paper in its entirety as it is the best organized formal description of the mathematics of regenerative response in a sound system that we are aware of.

To understand the transient behaviour of the feedback system, we must examine more carefully the process by which the steady-state condition, previously considered in the simplified analysis in section 2.1, is attained. This will be done following a method outlined by Antman[8] as follows:

8. Antman, H. S., " Extension to the Theory of Howlback in Reverberant Rooms " *J. Acoust. Soc. Am.*, Vol. 39, No. 2, February 1966, p. 399, (Letters).

2.2.1 *Build-Up Period*

Referring again to fig. 1, let p_1, as before, be the input pressure at the microphone. Assuming ideal transient transmission through the amplifier of gain μ, the instantaneous rise of pressure at the loudspeaker output is μp_1. Then, at a time $\Delta t = d/c$ later, an increment of pressure, $\mu\beta p_1$, is added to the input to give a new effective input, equal to $p_1 + \mu\beta p_1 = p_1(1 + \mu\beta)$, after one trip around the loop. Hence, a new output, $\mu p_1(1 + \mu\beta)$, instantaneously appears at the loudspeaker. Then, at a time $\Delta t = d/c$ later (or total elapsed time $2\Delta t$), the portion of the new output fed back to the microphone is $\mu\beta p_1(1 + \mu\beta)$. This combines with the original input p_1, which is assumed to be still present, to give a new input, after 2 trips, of $p_1 + \mu\beta p_1(1 + \mu\beta)$ or $p_1[1 + \mu\beta + (\mu\beta)^2]$.

Thus, after N trips, the input will be $p_1[1 + \mu\beta + (\mu\beta)^2 + \ldots (\mu\beta)^N]$ or $p_1[1 + \mu\beta + (\mu\beta)^2 + \ldots (\mu\beta)^{n-1}]$,

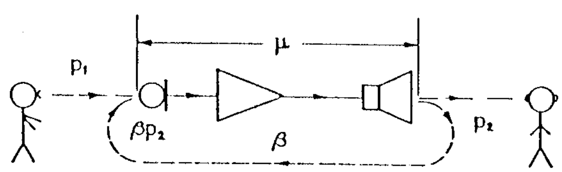


Figure 1.—Schematic diagram of an elementary sound system with an acoustic feedback path.

where n is the number of terms in the series and $N = n - 1$ is the number of trips. The resulting instantaneous output after a time interval $N\Delta t = (n - 1)\Delta t$ is thus

$$S_N = \mu p_1[1 + \mu\beta + (\mu\beta)^2 + \ldots (\mu\beta)^{n-1}]. \tag{9}$$

When $\mu\beta < 1$, the series converges to

$$S_N = \mu p_1 \cdot \frac{1 - (\mu\beta)^n}{1 - \mu\beta}, \tag{10}$$

and finally, as N, and therefore n, $\rightarrow \infty$, to

$$S_\infty = \frac{\mu p_1}{1 - \mu\beta} = p_2. \tag{11}$$

The ratio $\quad \dfrac{S_\infty}{p_1} = \dfrac{p_2}{p_1} = \dfrac{\mu}{1 - \mu\beta} = \mu'$

will be recognised as the steady-state system gain given in equation 2.

The system gain at $t_N = N\Delta t$, that is, after N trips around the loop, may then be written as

$$\mu'_N = \frac{S_N}{p_1} = \mu \cdot \frac{1 - (\mu\beta)^n}{1 - \mu\beta}, \tag{12}$$

$$= \frac{p_2}{p_1} \cdot [1 - (\mu\beta)^n], \tag{13}$$

$$= \mu'[1 - (\mu\beta)^n]. \tag{14}$$

Putting $G = |\mu\beta|$, as in equation 3, we may write

$$(\mu\beta)^n = G^n \epsilon^{jn\theta}. \tag{15}$$

Hence $\quad \dfrac{\mu'_N}{\mu'} = 1 - G^n \epsilon^{jn\theta},$

$$= 1 - G^n(\cos n\theta + j \sin n\theta),$$

$$= (1 - G^n \cos n\theta) - jG^n \sin n\theta,$$

$$= \sqrt{1 - 2G^n \cos n\theta + G^{2n}} \cdot \epsilon^{j\phi_N}, \tag{16}$$

where

$$\tan \phi_N = \frac{-G^n \sin n\theta}{1 - G^n \cos n\theta}. \tag{17}$$

Equation 16 gives the ratio of the system gain, after N trips around the loop, to the final steady-state system gain with feedback. To express μ'_N in terms of the gain without feedback, we have, from equations 4, 5, 6 and 16,

$$\frac{\mu'_N}{\mu} = \frac{\mu'_N}{\mu'} \cdot \frac{\mu'}{\mu} = \sqrt{\frac{1 - 2G^n \cos n\theta + G^{2n}}{1 - 2G \cos \theta + G^2}} \cdot \epsilon^{j(\phi_N + \phi)}. \tag{18}$$

This reduces, for $\theta = 0$, to

$$\left[\frac{\mu'_N}{\mu}\right]_{\theta=0} = \frac{S_N}{\mu p_1} = \sqrt{\frac{1 - 2G^n + G^{2n}}{1 - 2G + G^2}},$$

$$= \frac{1 - G^n}{1 - G} = \frac{1 - G^{N+1}}{1 - G}, \tag{19}$$

giving the gain ratio, μ'_N/μ, or the output amplitude ratio, $S_N/\mu p_1$, after a time $t_N = N\Delta t = (n - 1)\Delta t$ (note that equation 19 may be derived directly from equations 12 and 15 when $\theta = 0$). Values of $[\mu'_N/\mu]_{\theta=0}$ or $[S_N/\mu p_1]_{\theta=0}$ calculated from equation 19 for the first nine trips ($n = 9 + 1$) with $G = 0.6$ are shown in the left hand half of fig. 4. It will be noted that because of the time delay, Δt, in the acoustic path, which occurs for each trip around the loop, the output increases in a series of discrete steps, each containing $\Delta t/(2\pi/\omega)$ ($= d/\lambda$) cycles of the signal frequency.

Continued next page ...

In the steady-state condition, as we saw earlier, the input signal comprises two components. One of these is the sustained input p_1 ; the other is the fraction, β, of the final output, p_2 , which, having left the loudspeaker, arrives at the input with an amplitude βp_2 and phase shift θ (or time delay $\Delta t = \theta/\omega = d/c$). The decay of the system upon removal of the input may now be studied as follows.

2.2.2 *Decay Period*

When the input component, p_1 , is removed, the output drops suddenly by an amount μp_1 , corresponding to $x_0 - x_1$ in fig. 4, and the input continues to be supplied by the feedback component βp_2 which had left the loudspeaker before p_1 was removed. This continues for a time Δt during which the output remains at the level $p_2 - \mu p_1 = \mu\beta p_2 = X_1$ (say), (corresponding to $x_1 = \mu\beta x_0$ in fig. 4) ; the equivalent of $p_2 - \mu p_1$ and $\mu\beta p_2$ is seen from equation 1. At the end of the interval, Δt, the input suddenly drops from βp_2 to $\beta X_1 = \mu\beta^2 p_2$, resulting in a new output, $X_2 = \mu^2\beta^2 p_2$, which persists again for a time Δt. Thus, after N trips, the output will have fallen in a series of steps of duration Δt to a value

$$X_N = (\mu\beta)^N p_2 . \tag{20}$$

Normalising the output to μp_1 , we may write

$$x_0 = p_2/\mu p_1 ,$$
$$x_N = X_N/\mu p_1 .$$

and

Then, from equation 20,

$$\begin{aligned} x_N &= \frac{X_N}{\mu p_1} = (\mu\beta)^N \cdot \frac{p_2}{\mu p_1} , \\ &= (\mu\beta)^N x_0 , \\ &= \frac{(\mu\beta)^N}{1-\mu\beta} . \end{aligned} \tag{21}$$

Hence x_N gives the output amplitude ratio after N intervals of the decay period.

Values of x_N , which remain constant for each successive interval Δt, are plotted for $G = 0.6$ in fig. 4. The envelope is again a series of steps and has a form inverse to that of the build-up period. The amplitude ratio between adjacent steps is given by

$$\frac{x_N}{x_{N-1}} = \frac{(\mu\beta)^N}{(\mu\beta)^{N-1}} = \mu\beta . \tag{22}$$

Experimental confirmation of the response envelope shapes of fig. 4 is given by the photographs of fig. 5. These are single-trace records of the output of a simple feedback system fed with a short burst of a pure tone. The system had a predominantly single path of an integral number of wavelengths in length, ($\theta = 2n\pi$), and the gain was adjusted for values of $G = |\mu\beta|$ which may be shown, using equation 22, to be 0.74 for fig. 5(a) and 0.55 for fig. 5(b).

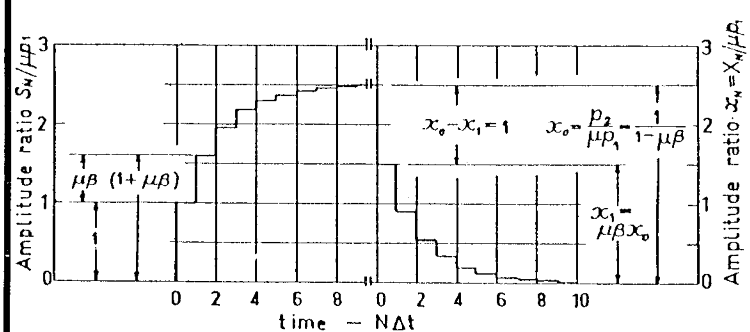


Figure 4.—Theoretical envelope of attack and decay responses of a system calculated from equations 19 and 21 for $G = 0.6$ and $\theta = 0$ (note that the steady-state gain ratio is 2.5, or nearly 8 dB as given in figs. 2 and 3).

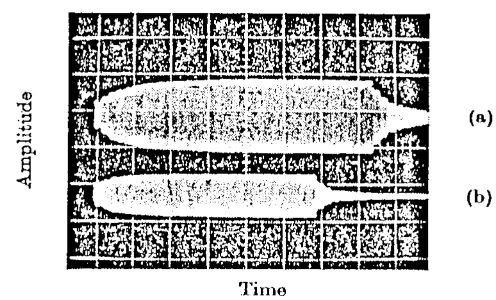


Figure 5.—Un-retouched photographs showing typical outputs of an experimental system with one predominant acoustic feedback path, for an input frequency for which $\theta = 0$.
(a) G = 0.74.
(b) G = 0.55. ♦

ACCURACY OF IMPEDANCE MEASUREMENTS

The problem of sufficient frequency resolution and amplitude accuracy when measuring the impedance of modern low frequency loudspeakers requires extremely narrow bandwidths and very slow sweep rates (i.e., on TEFs, FFTs, RTAs, etc.).

When using the "constant current" technique with our Crown RTA, we have found that the "pink noise" signal serves to rapidly locate the area of each resonant peak. Then connect an oscillator (the Loftech is ideal because it has a frequency meter) and tune for maximum level at each resonant peak or other area of interest. The levels reached with the oscillator will normally be higher than with noise. Convert the dB differences in the usual manner.

$$Z = \text{ref } Z \left(10^{\left(\frac{+dB}{20}\right)}\right)$$

The accuracy obtainable with the Crown RTA is comparable to that achieved with the TEF® so far as impedance magnitude measurements are concerned. ♦

ONCE AGAIN – HOW TO READ A VI INSTRUMENT

If it seems we're a stuck record on this subject, it's because of its importance to designers and installers. While operators of systems may never learn or need to learn what a level is and can function effectively on a reference reading, sound system installers and designers have the responsibility of setting that reference value correctly.

The operator of a system is given an instrument indication to use as upper limit to the electrical gain of the system, i.e., a zero instrument indication.

The designer and installer need to know both the "apparent level", i.e., the instrument indication plus the instrument attenuator setting, and the "true level", i.e., the apparent level corrected for any impedance difference between the circuit the instrument is across and the specified impedance of the instrument circuit itself. ♦

DOUBLE RESOLUTION ETC MEASUREMENTS

Charles Bilello showed all of us just how unobservant we could be during the recent TEF® Workshop. It turns out that he has been sweeping from -31,778 Hz to +31,781 Hz for a total bandpass of 63,559 Hz. That translates into:

$$\text{Full screen time} = \left(\frac{1}{\text{sweep range in Hz}}\right) 400 \quad \text{or} \quad 400\left(\frac{1}{63{,}559}\right) = 0.00629 \quad \text{or} \quad 6.3 \text{ msecs}$$

The resolution per line becomes:

$$\frac{0.00629}{400} = 0.000016 \text{ secs} \quad \text{or} \quad 16 \text{ usecs}$$

$$\text{Sound travels: } \frac{1130'}{1 \text{ sec}} \qquad \text{So: } \frac{1130'}{1 \text{ sec}} = \frac{x'}{0.000016 \text{ secs}}$$

$$x = 1130 \times 0.000016 \text{ secs} = 0.018'$$

$$\text{And: } 12(0.018) = 0.22 \text{ inches}$$

Or approximately 7/32nds of an inch.

In fact, this resolution was sufficient to produce the ETC of two misaligned loudspeakers shown in Figure #1 and, by merely taking the difference in cursor readings for both units (i.e., 0.54 msec) and dialing that into the Sunn ADS unit, the ETC of Figure #2 was obtained. Figures #3 and #4 show the before and after EFC curves (i.e., frequency response).

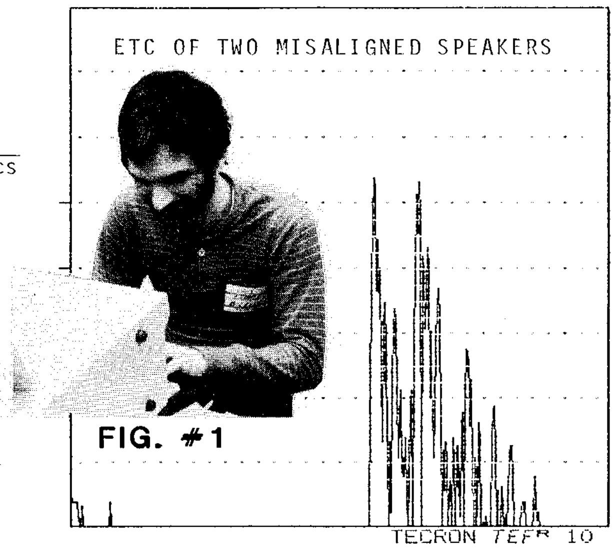


FIG. #1

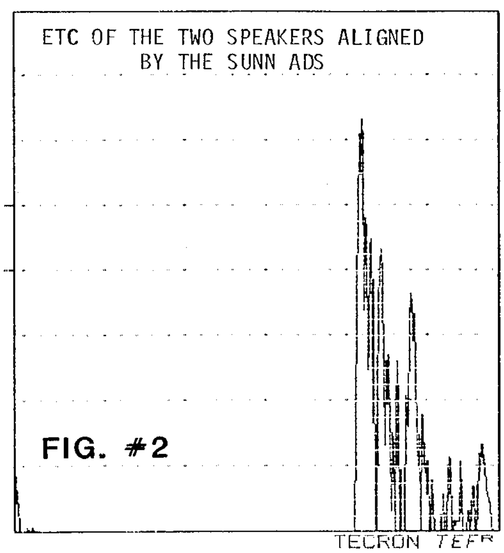


FIG. #2

TEF® SETTINGS FOR FIG. #1 AND FIG. #2

Vertical: 6dB/div with base of display at 36.1dB
0dB is located at .00002 Pascals

Horizontal: 0 microseconds or 0 Feet to
6278 microseconds or 7.09364 Feet
scale: 1.9395E+00 Feet/inch or 7.6357E-01 Feet/cm.
1716 microseconds/inch or 675 microseconds/cm.

Line Spacing: 15.7332 microseconds or 1.77785E-2 Feet
Line Width: 21.3971 microseconds or 2.41787E-2 Feet

Sweep rate: 5009.55Hz/Sec

Sweep range: -31778.90Hz to 31781.00Hz

Window file name: A:HAMMING.W8T

Input configuration: Non-inverting
with 24dB of input gain & 12dB of IF gain.

Continued next page.....

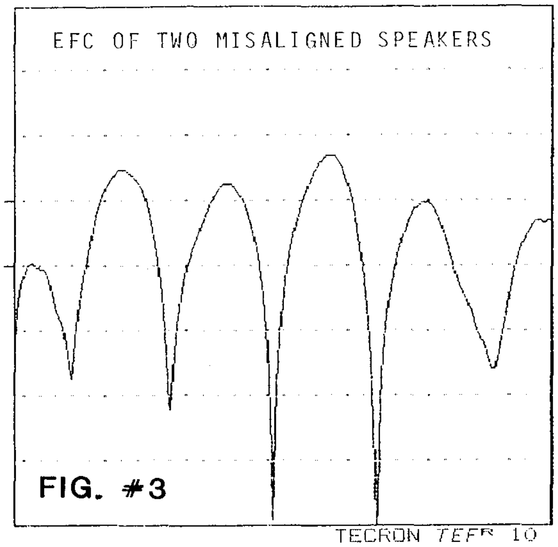


FIG. #3

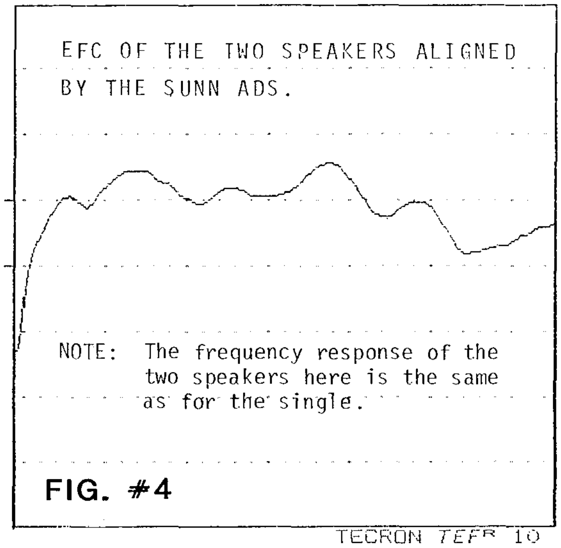


FIG. #4

TEF® SETTINGS FOR FIG. #3 AND FIG. #4

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

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

Resolution: 2.2557E+00 Feet & 5.0096E+02Hz

Time of test: 3477 microseconds, 3.9290E+00 Feet

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 1.0000E+01Hz

Input configuration: Non-inverting
with 24dB of input gain & 12dB of IF gain.

Note on the ETCs that we could not have been very far off of an exact alignment as the second ETC is +6dB above the first.

Note also the increased ability to see the internal reflections inside each loudspeaker enclosure. (Fig. 1 & 2)

Our congratulations to Charles Bilello and his wife, Kathy, who he credits with asking, "Charles, why can't you?" ♦

SAN ANTONIO - NOV. 21-22, 1983

ERRATA

An Important Correction

We wrote this important equation correctly clear back in Newsletter Volume 8, Number 4, page 27, Summer 1981. Rewriting it in the last Newsletter, Volume 11, Number 2, page 29, Winter 1984, we managed to thoroughly mess it up.

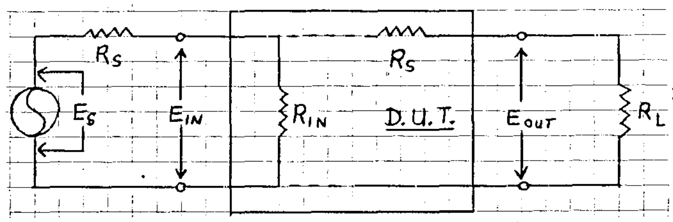
The lower right hand illustration on page 29, "Calculating the Insertion Gain or Loss of a D.U.T.", the equation for "G" should have read as shown here:

This is a fundamentally useful equation for circuit designers who would like to look at the input of a system (E_{IN}, R_S and R_{IN}) and the output of a system (E_{OUT}, R_L) as if it were a "Black box." This equation can be used either for a device or for a system.

We have also corrected and reproduce below the illustration which was found on page 28 at the bottom of the page, "Establishing A 'Sending Impedance' and Calculating The Available Input Power Associated With It."

This shows the breakdown of the theoretical AIP plus any insertion loss.

CALCULATING THE INSERTION GAIN OR LOSS OF A D.U.T.



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

*G is the "insertion gain or loss."

Calculating the Available Input Power (AIP)

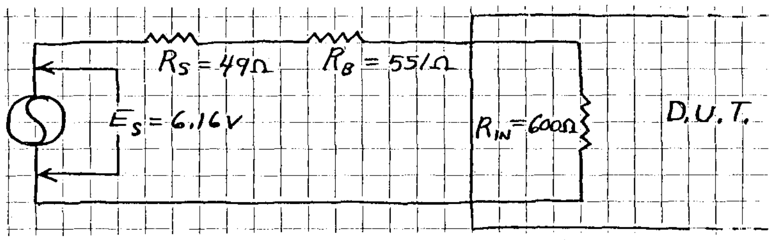
$$\text{AIP} = 10\ \text{Log}\left(\frac{(E_{out})^2}{0.001\ R_L}\right) - G$$

ESTABLISHING A "SENDING IMPEDANCE" AND CALCULATING THE AVAILABLE INPUT POWER ASSOCIATED WITH IT

Example

$$\text{Calculated AIP} = 10\ \text{Log}\left(\frac{(E_S)^2}{0.001(R_S + R_B)}\right) - 6.02\ \text{dB}$$

$$10\ \text{Log}\left(\frac{(6.61)^2}{0.001(600)}\right) - 6.02\ \text{dB} = 11.98\ \text{dBm}$$



Speaker/Boundary Errata

Syn-Aud-Con and Russ Berger wish to thank John Murray of Panacom, Dayton, Ohio, for bringing to our attention the following printing errors in Tech Topic, Volume 11, Number 5, "Speaker/Boundary Interference Response (SBIR)":

Example 2 should have stated in paragraph 3: "From this we know that the speaker should be mounted no *farther from* the boundary than 1.41 feet in order to avoid reduction of the power response. See Fig. 5."

John goes on to say:

> As can be seen in the first graph of Fig. 1, the reduction of power response occurs at wavelength distances between .2 and .5 wavelengths. Also, I feel its worth noting that in the case of two and three boundary situations, (Fig. 1, second and third graphs), the distance should be less than .15 wavelengths to retain the boundaries' response reinforcement. A ¼ wavelength distance, in these cases would be on or near the point of *maximum* cancellation, and therefore the least desirable position for the speaker.
>
> Example #3 states: 'The *sub-bass* speaker can, therefore, be no closer to the boundary than 4.52 ft. and the *mid-bass* speaker no further than 2.26 ft.' This should have sub-bass and mid bass reversed to compliment Fig. 4's example 3/section drawing. ♦

THE JOY OF SYN-AUD-CON

THE JOY OF SYN-AUD-CON IS THAT IT EXISTS AS A FORUM FOR
THE PRESENTATION OF NEW, OLD, ODD AND OUTSTANDING IDEAS
THAT NEED THE LIGHT OF SHARED AND FOCUSED MENTAL ENERGY.

QUESTIONS ABOUT CROSSOVER NETWORKS

TEF® analysis leads to a series of important questions when considering a crossover network for a loudspeaker system. Here are just a few of the new ones.

1. Should the crossover frequency be -3 dB or -6 dB when the two drivers are phase coherent?
2. What is the *acoustic* phase response of the system, i.e., the network plus the drivers as viewed at the listener?
3. In frequency time curves (FTC), if the "peak areas" are all at the same relative delay, are the differing decay rates at lower frequencies (A) audible (B) detrimental?
4. At the crossover frequency, should the levels be matched on a sensitivity measurement or on a power measurement in a reverberant space?
5. Has the selected driver and its directional control device (DCD) been studied as to the inversion of its coverage angles at some frequency within the range the network will supply to it?

These are but a very few of the questions that can be asked about crossovers.

Equalizers And Crossovers

One should avoid using any kind of equalizer in the crossover region as the slope rate of the crossover network filters can be dramatically altered by additional network elements. We would suggest, with the advent of TEF® analysis, that measurement of the acoustic phase response of the system as each equalizer adjustment is introduced will lead to a far clearer understanding of how to *properly* use an equalizer. ♦

MEASURING 70 VOLT LINE LEVELS

This is as straightforward a task as you are likely to find in audio; namely, what is the *level* of a 70 volt line?

1. Measure the impedance Z of the line.
2. Measure the voltage on the line (i.e., 70 volts or less).
3. Calculate (on the Syn-Aud-Con slide rule) the power and power level.

If desired, place a VI meter across the line with a suitable pad to maintain the impedance the meter is designed to see and adjust to a zero indication on the instrument at the level you desire as a reference level. The VI reading for VU is then the indication plus the pad value plus the impedance mismatch. ♦

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