SYNERGETIC AUDIO CONCEPTS

SYN AUD CON

newsletter

Volume 13, Number 1
Fall 1985

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

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

JBL'S NEW 5330 MICROPHONE MIXER

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JBL/UREI – A Bright Future

The JBL/UREI combination, along with the separation of the professional products into its own company responsible for its own development, is one of the important events of this decade. Commercial sound contractors have witnessed over the past twenty years the demise of the RCA Sound Divison, Stromberg Carlson, Altec, and many others, as conglomerates did what they do best - destroy small companies needing entrepreneural input. So, it is with great relief that we observe a major supplier of first rank products to commercial sound contractors being strengthened rather than destroyed. The personnel at the top at JBL Professional Sound Products, starting with the President, have had experience as sound contractors or in the sound contracting business. They have TEF capabilities. They are totally motivated toward the contractor's business. What more could an audio contractor ask of tomorrow's supplier. #

STAFF

Editors:
Don & Carolyn Davis

Design & Production:
Jan Kreitz
Pat Carlson
Debbie Lohrman

Syn-Aud-Con NEWSLETTERS and TECH TOPICS are published quarterly by Synergetic Audio Concepts, P. O. Box 669 (Rancho Carrillo), San Juan Capistrano, CA 92693.

This is Volume 13, Number 1. Fall 1985.

Subscriptions are available for $32.00 per year. Single or back issues are available. Write for price list. Air Mail subscriptions outside the United States are $38.00.

CRAFTSMANSHIP VS. ENGINEERING VS. SCIENCE

Technical endeavors divide into three major approaches. Pure science views the world as a transform of some sort either material or mental. Engineering looks at components and attempts to optimize each. On occasion engineers stray into systems work and leave tremendous legacies as a result, witness the dBm (now used in Fiber optics), the organization of impedance and admittance, and our entire diagramming techniques.

Craftsmanship is left to the workers who have to actually install, test, and operate the systems. There are famous schools of Science, major engineering universities, but craftsmanship is not taught. It is most often either inherited or found fresh by men and women who come across the work of a craftsman in the midst of an older system.

If you want to be a craftsman today, you'd better be in business for yourself. Business professionals usually want no part of such nonsense (except in the consumer products they personally purchase) in their own companies.

Craftsmanship springs from integrity and honesty in the individual. A craftsman is constantly seeking improvements (real improvements, not just new products), new knowledge, and improved skills.

Finally, a craftsman is known by his tools--not the number and expense of them--but how he or she uses them.

Craftsmanship is often mistakenly associated with the arts and crafts. The kind of craftsmanship discussed here is what we do in our daily work, not some hobby. Craftsmanship that is real is efficient, profitable and morale building, not a retreat to an outgrown or outmoded construction or use of an historical artifact. Craftsmanship is here and now, a pride, yea a compulsion to perform to the highest level possible for no material reason whatsoever.

A craftsman with a VOM will outperform the greedy, the falsely ambitious, and the big time operators, even if they have unlimited instrumentation. Where the *garbage* comes from in the saying GIGO is from the mind of man. Craftsmanship is the outward manifestation of an inner ear listening to the still small voice of conscience. #

NEW TEF-12 - $9,950

Tecron is now Techron. The TEF System 10 is now the TEF System 12. $16,000 is now $9,950. New literature flows forth, new people hard at work. Just a sample of what's happening is the release printed here.

We are reminded of the early days of Acousta-Voicing when we first had the 1/3 octave real time analyzer. We had the filters and we had the analyzers. What we didn't have was the organized sound system design techniques, but we generated them with a few years.

We now have the new analyzer. We have a proliferation of new devices that delay signals in various ways. What we now need is an organized sound system analysis technique that, once passed, truly insures that the sound system is operable by the end user. That comes from those of you in the field. Sharing of successful ideas, approaches, and even failures quickly point the way to improved methods.

The next few years will be fascinating as anything wrong in the old is replaced by the better in the new.

PRELIMINARY

TECHRON division of Crown International, Inc. TEF

SOFTWARE

WORKBENCH INSTRUMENT SOFTWARE VERSION 1.0

The Workbench System provides four often needed instruments for bench and field testing of sound systems, electronic equipment, and other electro-acoustic systems. These instruments include:

Sound Level Meter
Digital Volt Meter
Oscilloscope
Function Generator

These four instruments are all implemented on the TEF machine in a very user-friendly, easy-to-use manner. Pull down menus are used to allow instant access to any of the system's parameters and setups with full help facilities available.

Functions are provided to allow: storage of instrument setups and defaults, multiple printing options, selection of input options and instrument settings.

SOUND LEVEL METER

1 The sound level meter is displayed on the screen as two bar indicators calibrated in dB SPL showing respectively the processed detector output and maximum levels of the measured sound pressure.

The sound level meter system allows selection of several parameters including:

Frequency Response Weighting: Linear, "A", "B", "C"

Meter Top of Scale: 40 to 130 dB SPL

Meter Range: 10 to 60 dB

Meter Damping: Fast, Medium or Slow with separate settings for attack and decay time

Detector Type: Average, Absolute Peak, "+" Peak, "−" Peak, and RMS

Peak Hold: Track, Track Auto Release, Release

Calibration Constants: User Reference Units, Volts per Reference Unit, Zero dB SPL Reference Value

DIGITAL VOLTMETER

2 The digital voltmeter indicates voltage in both digital and analog formats. The measured voltage is shown in numerals and in a linear bar graph form where trends are easier to follow. Two bar graphs are displayed indicating processed detector output and maximum values.

The digital voltmeter has the following controllable parameters:

Function: DC, AC Average, AC Peak, and Impedance. The impedance function allows measurement of both the magnitude and phase (or real and imaginary) components of impedance at any frequency)

Meter Full Scale: 2 mVolts to 2 Volts (0.2 Volts to 200 Volts with external attenuator)

Meter Damping: Same as sound level meter

DIGITAL OSCILLOSCOPE

3 The oscilloscope indicates the time history of the input voltage on a displayed graph on the screen. The scope will run in the triggered or free run sweep modes. The following oscilloscope parameters can be selected:

Horizontal Sweep Time: 500 uSecs to 2 Secs per Div

Vertical Sensitivity: 1 mVolt to 1 Volt per Div. (0.1 Volts to 100 Volts with external attenuator)

Sweep Trigger: Free Run, Source, Keyboard

FUNCTION GENERATOR

4 The function generator can generate a number of continuous signals including: fixed frequency, swept frequency, and random noise.

The function generator runs independently of the other instruments in the Workbench package. This means that the generator can be used as a test stimulus for the other instruments such as the sound level meter, voltmeter, and oscilloscope. The following settings can be selected:

Function: Sine, Triangle, White Noise, Pink Noise

Mode: Fixed Frequency, Swept Frequency (sweep can be started, stopped, paused, and continued)

Output Voltage: 10 mVolts to 2 Volts RMS

Sweep Time: 0.5 Secs to 50 Secs

Frequency Entry: Direct Keyboard Entry, One-third-octave selection, Increment-Decrement in linear and log steps.

SYSTEM FUNCTIONS

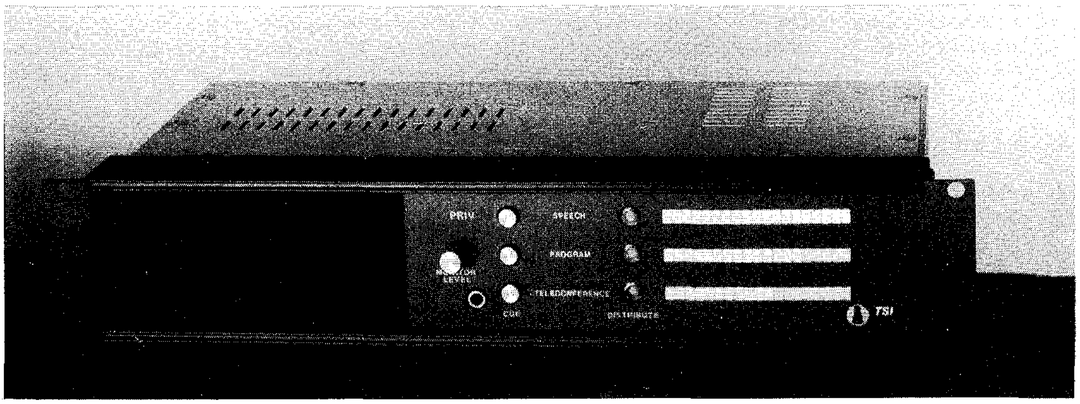
5 The workbench system has the ability to store and retrieve instrument setups including a user defined default setup that defines all the instrument settings on system startup. Menu selections are provided for quick return of each instrument to its default setup (both collectively and individually).

The user is provided multiple print options, easy to access help information, and a variety of input options including balanced-unbalanced, inverting etc. to make the instruments as easy to use as possible.

AUDIO MONITORING & INDICATOR PANEL

Mario Maltese of TSI in Mineola, New York, started attending Syn-Aud-Con classes in 1974. And we see him every few years in a Syn-Aud-Con class.

This year Mario, Chris Maione and Brett Sandgren from TSI brought into the New York class a product they have developed:



FEATURES

- Visual and aural monitoring of up to three channels of audio sources in 3½" panel space.
- Custom engraving on front panel for each channel (i.e., speech, program, teleconferencing).
- Pop and click protection of distributed loudspeakers using muting relay with two second delay at turn-on and instant off at turn-off.
- Distribute Switch: in "off" position, operator can aurally preview each channel before distribution to loudspeakers; in "on" position, program is distributed to loudspeakers.
- Isolated cueing of each channel with a cue or solo momentary pushbutton.
- Separate microphone input with phantom power for monitoring oral cues. (Microphone not supplied.)
- Separate auxiliary input for testing and utility functions.
- Privacy Feature: closure activated mute circuit defeats monitor speaker while continuing to visually display volume level.
- Line output for recording.
- DC rails at rear connectors (+5 volts, +15 volts, −15 volts).
- Easy to use captive screw connectors on rear panel.

We were so impressed with the product that we asked Mario if we could write it up and make a mailing in our Newsletter so our "grads" would know about it. (Mario is producing it in quantity for about $800 - but get the details from Mario.)

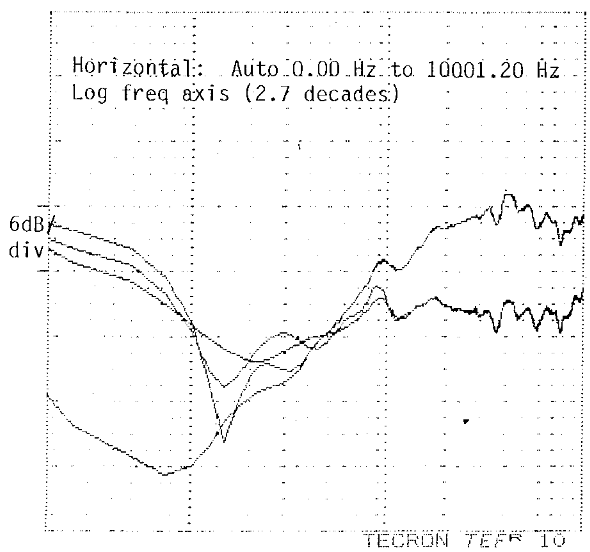
The MVI Series Amplified Monitor Panel is a must for controlling the sound system from the equipment rack location, which is often in a different acoustic environment than where the distributed loudspeakers are placed. It's a high impedance device that will accept anything from "aux" level to 70 volts, and it packs quite a punch for its size.

After years of hand-wiring many of the same features into our corporate and theater audio/visual-video systems, we decided to produce a standard quality product and clear our racks of many relays, switches, L-pads that burn out, ugly meters and range switches, custom metalwork and wasted front panel space.

"SMART" AMPLIFIER ADAPTER?

We recently had an occasion to measure an expansive tone control masquerading as a "smart" amplifier adapter, only it went in front of the amplifier, not on its output.

We'll let the curves speak for themselves.



SMILE

The most fertile source of insight is hindsight. #

A STANDARD FOR LOUDSPEAKER TEST

The argument is occasionally put forth that a loudspeaker measures well but sounds poor. Indeed they can. What you really need to know, however, is why did they sound bad? The answer may be that they measured the wrong things or failed to measure *all* the necessary things. Just as prevalent is the excellent loudspeaker that sounds bad because the source material played over it is bad. It's in this area that the entire Hi-Fi underground receives its fertilization. Speaker A sounds great on program X. Speaker B sounds terrible on program X. Then a clean recording accidentally falls into their hands and now Speaker B is great and A is trash. Because of this problem, subjective testing should always and only be done by yourself.

Objective tests can be useful if the correct ones are chosen:

1. A steady state frequency response of the direct sound level with an F_R as small a band as the test space allows.

2. A detailed energy time curve (i.e., -31669 to +31669 with a 4 x expansion).

3. Both linear frequency and logarithm frequency response is useful. I like the linear in three separate decades: 0-200 Hz, 200 - 2000 Hz, and 2000 - 20,000 Hz. Side-by-side views of Log-Lin sweeps are most educational. For Log sweeps use the Tecron 1.1 disc *not* the Heyser disc, because you can then *start* at any desired low frequency and still have an accurate screen scale. On Heyser's disc, in order for the log scaling to be correct, you need to always start at 0 Hz.

4. A 3-D response adjusted to allow the behavior of the decay side of the response to be seen in reasonable detail (i.e., Fs = 250 Hz or thereabouts with T_R adjusted to fill the screen from T_1 to T_2).

5. The Nyquist polar plot.

6. The phase vs frequency plot.

7. A 3-D polar response of both vertical and horizontal directivity.

8. A "buzz" test for audible buzzes, rattles, voice coil rubs, etc.

9. A sample distortion run a la Heyser can be included, but most good loudspeakers should barely appear on the screen.

My personal opinion is that I can pretty well judge what a loudspeaker is going to sound like from the first seven tests. The perpetual challenge is how to handle the hype that goes with special devices good on special sources. For instance, it's conceivable that some loudspeaker might "mask" the problems in some digital recordings. Does that make it a good loudspeaker? "Beauty may, indeed, be in the ear of the beholder."

Comments On The Above

Before we published the above, we sent it to Neil Grant in England, who will be soon writing speaker reviews for Studio Sound, and who will soon market a studio monitor in the United States through Bob Todrank at Valley Audio. We are printing excerpts from his letter:

> Define the external parameters......type, position and input to the amplifier. Location of the speaker and adjacent reflective surfaces. Use Dick's (Heyser) equation to predict the response ripple in this position. Standardise on the chosen set-up. (I have to agree with Peter Baxandall's comments about the similarity of amplifiers and the rubbish that is talked about interconnects, but I still feel that it is worth defining.)
>
> Modulus of impedance. If particularly complex, experience shows that it may be necessary to divide the bandwidth up in sections, very much as with the amplitude response family.
>
> Distortion. Here I disagree a little with Don. Fair enough, most good loudspeakers should measure well, but if they were all good ones, there would be little point in reviewing them all. Distortion is also interesting where it departs from the levels that you would anticipate, or where the device misbehaves momentarily. This can give insight to some of the other mechanisms colouring the response. My only concern is the display itself...if you use one of Dick's difference modes to enable reading relative levels instantly, the curve is relative to the fundamental, and misleading if printed in that form. Interpretation from the basic curve on its own can be difficult, unless, again, you divide up the frequency scales.

A BIT OF WISDOM

Better to light a candle than to curse the darkness. Gerald Stanley at the TEF Workshop #

1986 SCHEDULE

SOUND ENGINEERING SEMINARS

January 14-15 — Anaheim, CA
Quality Inn

March 26-27 — Portland, OR
Red Lion (Jantzen Beach)

April 2-3 — Vancouver, B.C.
Holiday Inn (Downtown)

April 24-25 — Las Vegas, NV
Hacienda Hotel

WORKSHOPS

February — What would you like?
West Coast

We would like to host a Workshop on the West Coast in February. What would you like? We have included a form for you to send us which will help us put it together.

Studio Designer's Workshop
May 7-9 — Dallas, TX
Staff:
Russell E. Berger
Peter D'Antonio
Doug Jones

SHURE ON THE MARCH AGAIN

Shure's FP 11 and FP 12 are two extremely useful audio system tools.

The FP 11 microphone-to-line level amplifier allows a microphone to be fed from a remote location over a balanced circuit at line level. We know of no better cure for overcoming noisy lighting controls. The precision 6 dB/step attenuater allows matching of incompatable levels from various sub mixers as well.

The FP 12 is a true bridging amplifier (very high impedance tap across lower impedance lines.) This unit literally allows you to bridge across any system link circuit and have a look without effecting the operation of the link circuit itself.

Both units are very reasonably priced with the FP 11 $175.00 and the FP 12 $160.00. Both are battery operated requiring a single nine volt battery. We are extremely interested in Shure's current approach to professional electronics and as the lessons from the field come in, we suspect we'll see a super interactive group of components that make complex systems much easier to tie together.

Battery test points ("Batt test") can be used to test the battery under load <u>or</u> to insert 9 - 12 volts to run the units. Use <u>a</u> well regulated and filtered supply. #

WHY SYN-AUD-CON?

Shallow thinkers are often tempted to put down Syn-Aud-Con's two-day seminars as not worthwhile because of their brevity. Indeed, who would not agree that you can't become an expert in audio in two days? What is overlooked is that you won't become an expert in audio from *anyone's* course no matter how lengthy and that those who attend the Syn-Aud-Con basic two-day classes are not raw beginners but men and women already employed in audio work who have sensed the inadequance of any form of study except continuing adult self-study.

For those truly desirous of pursuing an academic background of genuine use to a lifelong audio career a full university physics course is the correct path.

What a Syn-Aud-Con two-day seminar can do for those who attend is to *enable* them to identify those tools a real professional uses and needs and to learn where to acquire the self-learning materials necessary to the pursuit of this newly identified knowledge.

After 14 years of Syn-Aud-Con experience we know that what holds back people in audio is a failure to identify what they don't know and that they need to know. A Syn-Aud-Con two-day seminar is devoted to "washing" the attendee in the pool of audio technology and allowing them to self-identify their needs. The more experienced a person thinks they are the more beneficial the uncovering in their own mind of missed or misunderstood fundamentals. That's why those fortunate enough to have attended a Syn-Aud-Con class first know how to direct their continuing educational efforts with maximum effectiveness. #

THE SIGNIFICANCE OF 1/f IN ACOUSTIC MEASUREMENTS

The inverse of the concept of "frequency" ($1/f$) is the primitive period (i.e., one wavelength "λ"). As the words imply λ is a length.

$$d = 1/f \; c = \lambda$$

where: d is the distance in ft or M
f is the frequency in Hz
λ is the wavelength in Ft or M
c is the media velocity in ft/sec or M/sec

Measuring Low Frequencies

What are the necessary and sufficient conditions for measuring low frequencies either indoor, outdoor, anechoically, or otherwise? The answer is the development of one full wavelength.

This is one of the basic meanings of the time-bandwidth product

$$F_R \bullet T_R = 1.0 \qquad \text{or } T_R = \frac{1}{T_R} \qquad \text{and } T_R = \frac{1}{F_R}$$

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

The Critical Frequency (f_C)

The term "critical frequency" f_C is used to describe the frequency below which standing waves cause significant room modes. This is a frequency usually quite close to the $\sqrt[3]{V}$ (i.e., $V^{(1/3)}$) as a wavelength.

It is apparent that genuine difficulties arise in acoustic measurements whenever you are at a frequency whose wavelength exceeds the dimensions of the enclosed space where the measurement is being made.

TEF® VS Anechoic Measurements

The lowest frequency you can measure accurately with high resolution requires the development of at least one full wavelength (i.e., with the TEF® analyzer set for a T_R that rejects any other signal). This is a requirement for *any* measurement, not just TEF® measurements.

It is then, we hope, obvious that with the TEF® analyzer all that is necessary is sufficient space around the device under test (D.U.T.) and the measurement microphone. Since this is true for the so called anechoic chamber, it then becomes *a totally unneeded tool* for two port measurements.

Those who do not understand this do not understand the fundamentals of any acoustic measurement and they should review their grasp of fundamentals before doing any acoustic measurement work.

Choosing The Lowest Frequency

From the above, it follows that the lowest frequency you should attempt to measure with high resolution is a function of the nearest reflecting surface. (Remember, in an anechoic chamber, they *are* reflective at low frequencies.) Thus:

$$f_{LF} = \frac{c}{d}$$

where: f_{LF} is the lowest frequency in Hz for a high resolution measurement
d is the distance in Ft or M between the L_D on the ETC and the first reflection.
(Note the first reflection may be associated with either the D.U.T. or the measuring microphone.) ♦

THE HEYSER TRANSFORM

The clearest public statement to date of the existence of the Heyser Transform is Dick Heyser's abstract for his paper given at the May 1985 AES Convention on the West Coast.

He states that this is "a new integral transform. It is not a case of the new transform versus the Fourier Transform: The new transform includes the Fourier Transform as a special degenerate case. Analysis properly performed under the Fourier Transform can be precisely duplicated under the new transform *but not conversely* (italics are mine)......The conversion between conventional time and frequency domains may now be accomplished by intermediate maps *through other domains* (italics are mine), reaching signal properties not available to the direct Fourier Transform map. *This includes nonlinear* (italics are mine) attributes such as wavelength-dependent dispersion and amplitude-dependent distortion."

Until Dick actually publishes the full mathematical treatment of his transform, this manifesto will have to suffice. Those of us employing this transform as embodied in the TEF® analyzer can see, feel, and hear the results. We suspect that full recognition of what has happened and what consequently can happen will take until the year 2000 AD to flower. (AD stands for "agonizing delay.") #

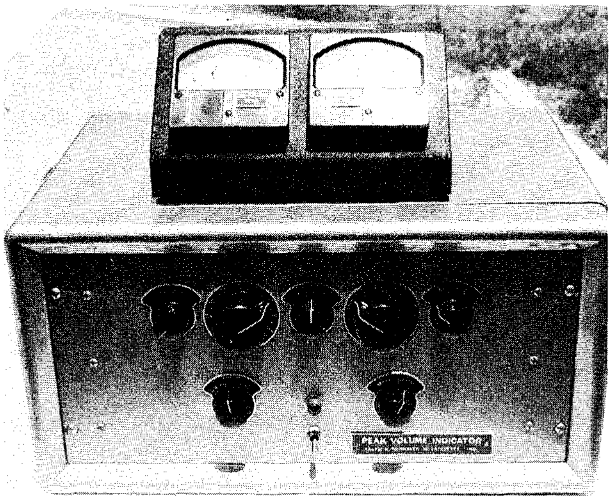
RALPH TOWNSLEY & HIS PEAK READING METER

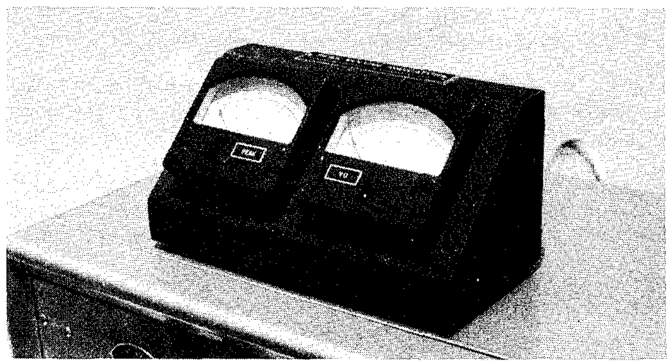
James Michener has written that a young boy is lucky to find a grownup to follow as a model.

One of the grownups I looked up to and still do is Ralph Townsley, who, when I was a boy, was the Chief Engineer of WBAA at Purdue University. He had literally hand built the station and his audio engineering ability (in fact, his pure engineering ability) has stood as a goal to strive toward for nearly fifty years of my life.

Recently, while visiting Ralph at his home in West Lafayette, IN, (the town I was raised in) I found that one of the two peak reading VI instruments he had designed and built over forty years ago could be purchased. I quickly agreed and now have the remarkable instrument shown in the photographs.

Ralph Townsley (L) with members of the Indianapolis class.





It is in perfect condition and operates flawlessly. Ralph completely reconditioned it, found new tubes for it and calibrated it. This instrument gives the normal VI reading in VU on one meter and the true peak value (in one cycle out of ten thousand) on the other. It can be set to hold the maximum peak, indicate maximum and slowly return, or indicate maximum and rapidly return.

We had used this instrument over thirty years ago to demonstrate the then new Altec condenser microphones to our Hi-Fi customers in Indiana. We discovered that a 200 watt tube type McIntosh gave enough headroom on a Klipsch horn to *never* clip and, as a consequence, generated less audible noise from records and tapes as the peak energy of the noise caused many of the lower powered amplifiers of that day to clip.

Ralph conceived of, hand built, and used this instrument to monitor his audio quality to the transmitter. (Harry Miyahira, President and owner of HME was privileged to work at WBAA as a young EE student at Purdue.)

Ralph Townsley is a quiet man, not given to promotion of himself. Men like Ralph seldom know the impact their character and integrity have on others. I wish every young man with an interest in audio could have a Ralph Townsley as his mentor.

BOSE PATENT

United States Patent 4,490,843 granted to Amar G. Bose and Richard G. Plourde describes a loudness compensation circuit that simply progressively emphasizes a peak centered at about 50 Hz and does nothing else to the remainder of the spectrum.

Increasing evidence is building that psychoacoustics allows usable results from relatively narrow band peaks *if* they are centered at the correct frequencies and levels.

The sweet sounds heard in a concert hall are the results of comb filtering. Those who choose sound production over sound reproduction should be able to find a significant marketplace for their acoustic illusions.#

CONCEPT OF POWERS

In dealing with the concept of powers, both electrical and acoustical, and their conversion into power levels, we need to keep the following basic relationships in mind.

Electrical Power (W_e)

Electrical power is measured in watts (W_e). The familiar ohms law form encountered in many textbooks is:

$$We = \frac{E^2}{R}$$

Where: W_e is the electrical power in watts

E is the electromotive force in volts

R is the resistance in ohms

Acoustical Power (W_a)

Acoustic power is also measured in watts:

$$W_a = \frac{(P_{RMS})^2(\text{area})}{pc}$$

Where: W_a is the acoustic power in watts

P_{RMS} is the sound pressure in pascals (pa) as a root mean square value

Area is the area around the source having the indicated P_{RMS}

pc is the acoustic characteristic resistance in RAYLS (406 RAYLS for air at sea level)

$$P_{RMS} = 2\Pi fApc$$

Where: f is the frequency in Hz

A is the RMS amplitude in M

Acoustic Intensity

Acoustic Intensity I_a is the *power per unit area* at some given observation point in watts per square meter (w/M^2).

$$I_a{*} = \frac{(P_{RMS})^2}{pc} \text{ (at a radius r)} \qquad {*}(I_a = 4\Pi^2 f^2 A^2 pc)$$

Levels in Audio and Acoustics

In our work in audio and acoustics, level is always *power* level. A voltage amplitude, a sound pressure amplitude, etc., are first converted into a power ratio equivalent and then that power ratio is converted into an appropriate level. These levels and their reference levels are listed below.

ELECTRICAL POWER LEVEL

$$10 \text{ Log}\left(\frac{W_e}{0.001w}\right) = L_{dBm}$$

ACOUSTICAL POWER LEVEL

$$10 \text{ Log}\left(\frac{W_a}{10^{-12}w}\right) = L_W$$

ACOUSTICAL INTENSITY LEVEL

$$10 \text{ Log}\left(\frac{I_a}{10^{-12}w/M^2}\right) = L_I$$

SOUND PRESSURE LEVEL

$$20 \text{ Log}\left(\frac{P_{RMS}}{0.00002 \text{ pA}}\right) = L_P \qquad L_P = 10 \text{ Log } \frac{(P_{RMS})^2}{(0.00002 \text{ pA})^2}$$

Thus, we have both powers and power levels. We can calculate power levels either from powers or from amplitudes by using the appropriate multipliers. In the latter case, the impedance of both the measurement and the reference must be the same.

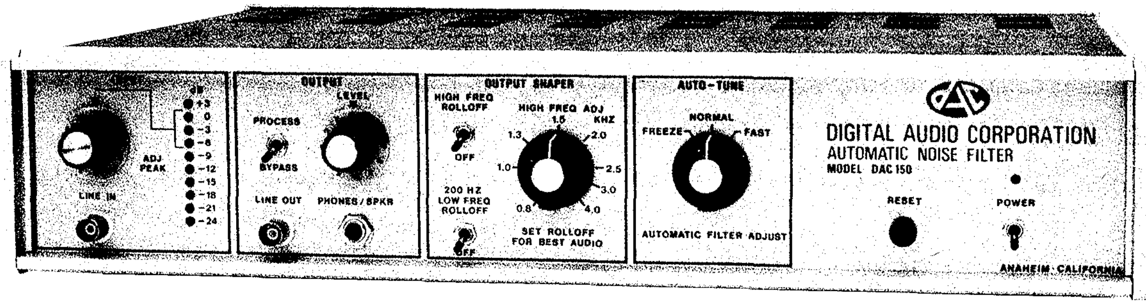
We hope these definitions, equations, and this discussion help you sort out the differences between these valuable parameters. #

ADAPTIVE PREDICTIVE DECONVOLUTION PROCESSORS

These processors have typically been in the $12,000 to $24,000 category in the past and are now approaching costs that can be considered in some special cases. We are reproducing below a portion of a specification sheet from one of the most advanced and reputable producers of such devices. Note carefully that the standard instruments are limited to 5 KHz and wider range units are a special order option.

It is our belief that somewhere in the not too distant future such processors will be both faster and cheaper, hence, used frequently.

DAC 150 AUTOMATIC NOISE FILTER



General Description

"The DAC 150 is a second-generation adaptive digital filter system intended for applications in automatic noise cancellation and voice intelligibility enhancement. This instrument is the fourth such filter introduced by Digital Audio Corporation and incorporates features heretofore available only in units costing two to four times as much.

The DAC 150 incorporates a large adaptive filter to provide precise noise cancellation automatically. Reverberation, power line buzz, room noises, acoustic resonances, spectral distortion, channel interferences, telephone line effects, music and general background noises are greatly reduced on voice signals without operator tuning. The DAC 150 may be applied in the laboratory to pre-recorded tapes, in the field to real-time voice communications, or in the studio to remote newscasts and interviews.

APPLICATIONS

VOICE TAPE RECORDINGS - Law enforcement and other voice recordings made under undesirable conditions may be greatly enhanced Body, transmitter, and room recordings often have disturbing effects such as room resonances, reverberations, restaurant noise, buzz, and air conditioner noise, to name a few. These noises are substantially reduced making the recordings more intelligible and listenable Transcriber throughput is increased and fatigue reduced Presentation to juries becomes more effective.

RADIO COMMUNICATIONS - The DAC150 is an automatic QRM filter. Band edge ringing from strong adjacent-channel signals and co-channel interference from FSK and CW are virtually eliminated. Multipath effects are convolutional in nature and are similarly reduced. The adaptive filter, furthermore, tracks the changing noise characteristics and provides optimal noise cancellation without degrading the voice quality.

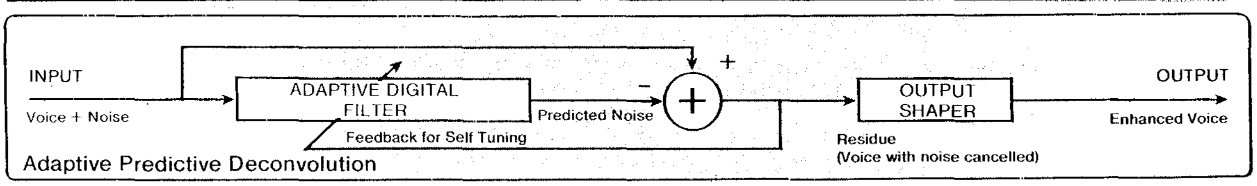
NEWS BROADCASTS - News recordings and interivews are often made under undesirable acoustic conditions. Background noises and acoustic effects may be removed by the DAC150, making these recordings more presentable for broadcast. Remote newscasting, via radio or telephone, often has telephone line effects, muffling, and helicopter and other noises which may be greatly reduced by a DAC150. The instrument may be left in the line as a filter and bypassed when cancellation is not desired.

INSTRUMENT OPERATION

The DAC150 is an adaptive digital filter system. The precise filtering is carried out by first converting the input audio signal into numeric (digital) form 12,500 times per second and storing these samples in an internal memory. The samples are arithmetically processed at a rate of 15 million operations per second by a special-purpose computer. Noise components are removed, leaving the residual voice samples intact. The residue samples are subsequently reconverted to audio form for output. The noise cancellation process carried out is adaptive predictive deconvolution and is illustrated in the figure below. A large digital filter predicts the signal slightly in the future, based on past samples stored in memory. The predicted samples are subsequently subtracted from the actual samples, leaving only the unpredictable signal components as the residue. The residue is, furthermore, fed back, forcing the filter to tune itself continuously for optimal prediction.

Additive noises, such as tones, buzz, and air conditioner noise, and convolutional noises, such as resonances, reverberations and acoustic effects, are very predictable and are cancelled by this process. Voice, however, is more dynamic in nature and is not predictable. The voice passes through the filter as the residue without cancellation or modification. Since the filter is continuously tuning itself, it will track and cancel slowly varying predictable noises such as music and non-stationary communications channel variations.

The DAC150 provides a NORMAL self-tuning speed for the majority of noise and a FAST for rapidly changing noises. A panel-adjustable output shaper allows the operator to reduce residual low-frequency and high-frequency noises.



Adaptive Predictive Deconvolution

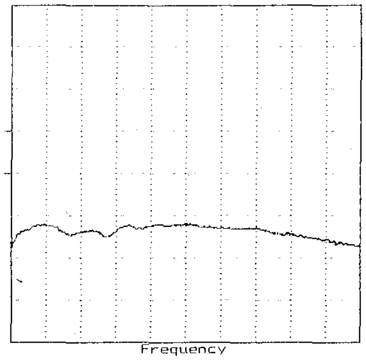
DIGITAL AUDIO CORPORATION • POST OFFICE BOX 51539 • RALEIGH, NORTH CAROLINA 27609

THE LOUDSPEAKER THAT WORKS - BUT!

During one of the "hands on" sessions at the Hamburg Super Workshop, one of Hellmuth Kolbe's Visonik David 6000s fell to the floor. It still worked when picked up and might have indeed been used by someone without test equipment -- **but** -- witness the before and after data.

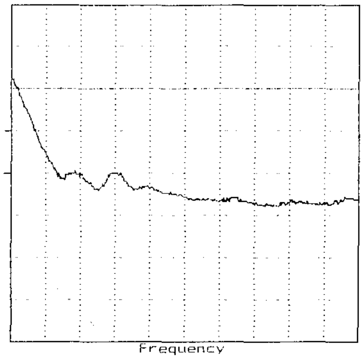
Hellmuth (C) with Richard Heyser (L) and Jim Brown (R).

FIG. 1 - SN 680528
Good Speaker



Vertical: 12dB/div with base of display at -115.8dB. OdB is located at .775 Volts.

FIG. 2 - SN 680528
Good Speaker



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

FIG. 3 - SN 680528
Good Speaker

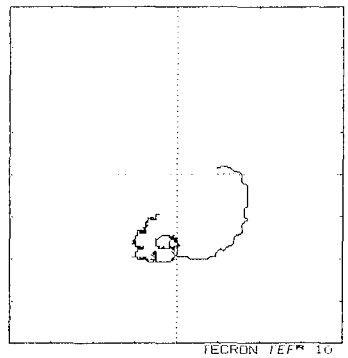
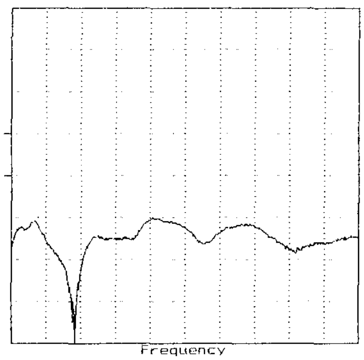
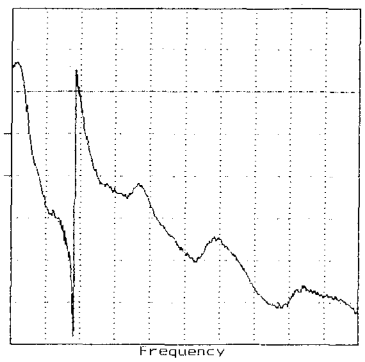


FIG. 4 - SN 680530
Retest After Drop



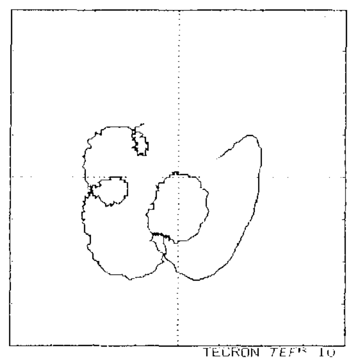
Vertical: Linear amplitude.

FIG. 5 - SN 680530
Retest After Drop



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

FIG. 6 - SN 680530
Retest After Drop



PARAMETERS FOR FIGURES #1, #2, #4 and #5

Horizontal: 20.35Hz to 19998.10Hz
scale: 5462.12Hz/inch or 2150.44Hz/cm.

Resolution: 3.1952E-01 meters & 1.0735E+03Hz

Time of test: 8731 microseconds, 2.9947E+00 meters

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

Input configuration: Channel 1 Non-inverting
with 48dB of input gain & 21dB of IF gain.

PARAMETERS FOR FIGURES #3 and #6

Resolution: 3.1952E-01 meters & 1.0735E+03Hz

30dB of automatic screen gain.

Frequency range: 20.35Hz to 19998.10Hz

Time of test: 8747 microseconds, 3.0002E+00 meters

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

Input configuration: Channel 1 Non-inverting
with 48dB of input gain & 21dB of IF gain.

SMILE

The bitterness of poor quality remains long after the sweetness of low price is forgotten.

Sent in by: Harvey Earp, J. W. Davis & Company.

SPEAKER DESIGN PHILOSOPHY

To be polite, we'll assume that what we are about to share is the result of differing philosophies and not accidents that occurred.

Figure 1a is the ETC of a loudspeaker built by a southern manufacturer for use by musical groups. Using the Heyser disc and an expansion of x4, we are able to see the signal misalignment between the woofer and the high frequency unit. What kind of problem does this cause? Figure 1b reveals the end result. Our comment- "Well, it's unlikely to 'feed-back' at that frequency." Figure 1c is the linear frequency scale reading of the same unit.

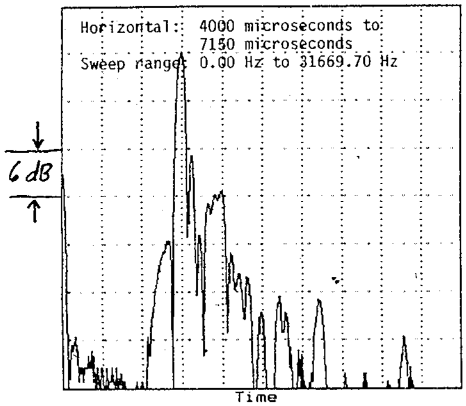
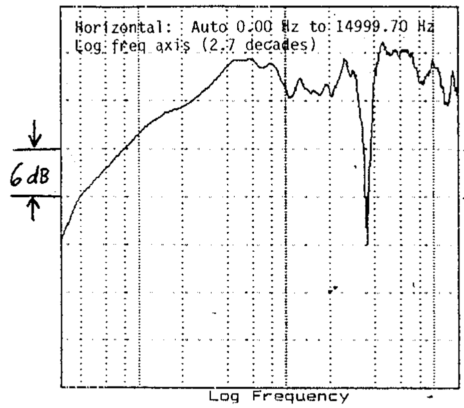
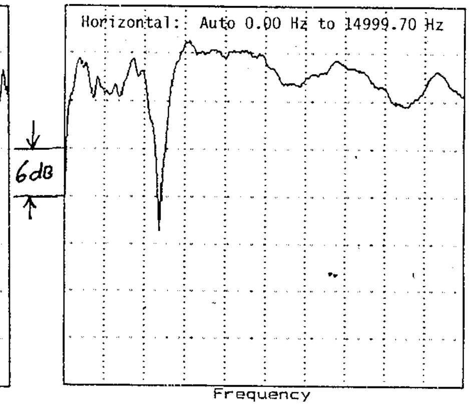


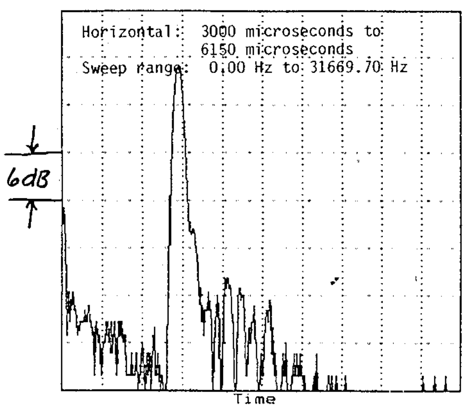
FIGURE 1a. A monitor for musical sound market (4 x expansion). Note severe misalignment.

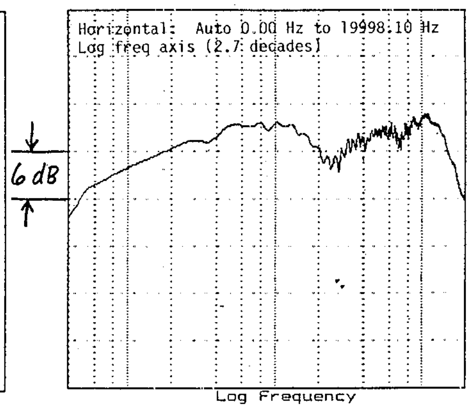


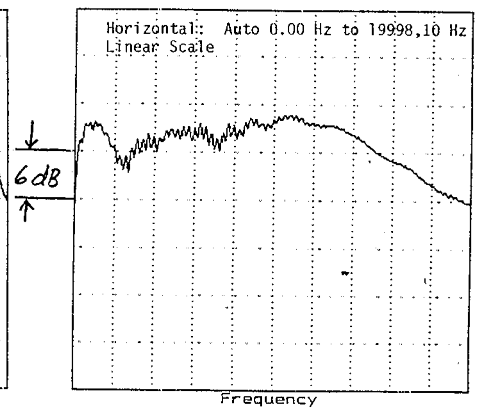


FIGURES 1b and 1c. Log & linear scales, showing the effect of the misalignment on the frequency response. The aberrations are every 3000 Hz, but the one comb filter fell in the crossover region -- always disastrous.

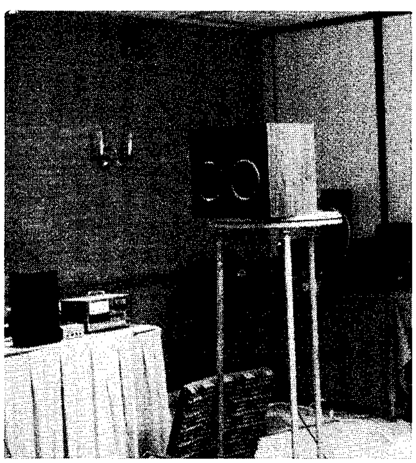
Figure 2a is an ETC of an all cone signal aligned system by a New England manufacturer. This system exhibits excellent control of internal reflections, alignment of drivers and networks. Figures 2b and 2c are made out to 20 KHz and reveal a ± 3dB behavior over most of the range.



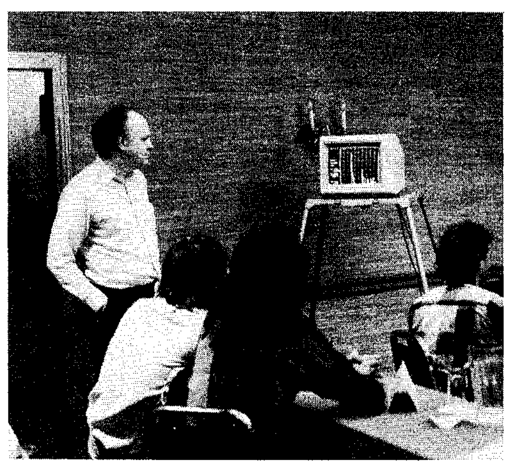




FIGURES 2a and 2b and 2c show Log and linear scales of an aligned loudspeaker system.

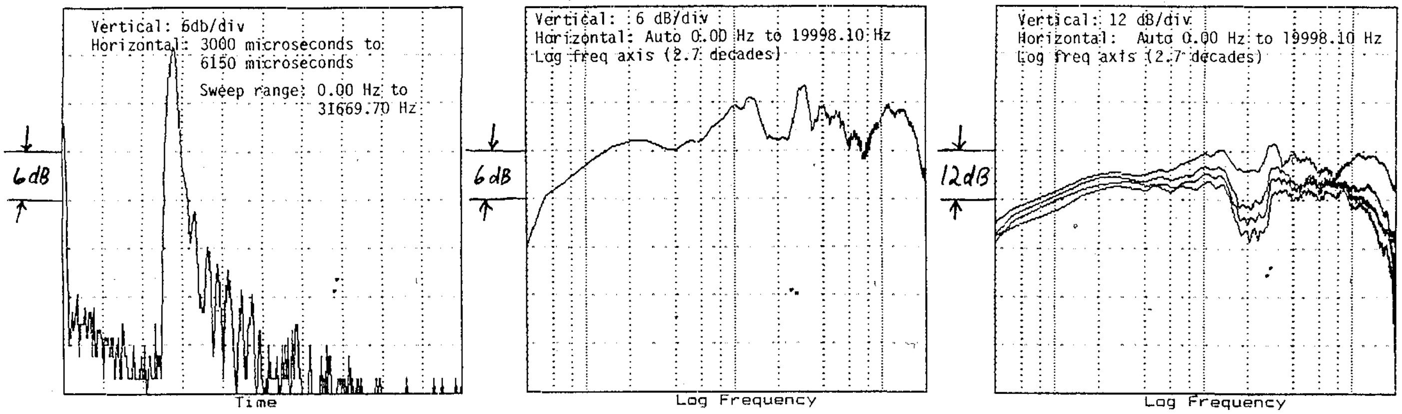


FIGURES 2d and 2e show the Eastern Acoustic Works loudspeaker and designer, Ken Forsythe, under test.



Continued...

Figure 3a is the ETC of a Japanese Manufacturer for a small near field type monitor. Figure 3b shows another version of ±3dB to 20 KHz (log scale). Figure 3c, however is the interesting one, revealing the on and off axis behavior to 30°.



FIGURES 3a, 3b, and 3c. A small "near field" monitor type loudspeaker. Note the exceptional off axis response. The 2000 Hz dip is probably crossover problems again.

Finally an ETC of an American manufacturer who simply does not believe in signal alignment. See Figure 4.

All of this data was gathered in a single Syn-Aud-Con basics class one evening. We tried all of these units on voice as well.

While not quarreling with any of these approaches by manufacturers, it was clearly manifest that the signal aligned units also sounded the clearest on voice.

The last few years have shown us that there are indeed two major schools of thought: one avoiding signal smearing and one cultivating it. We have strong personal preferences in this matter, but our real interest lies in what will be the final judgment in the marketplaces. #

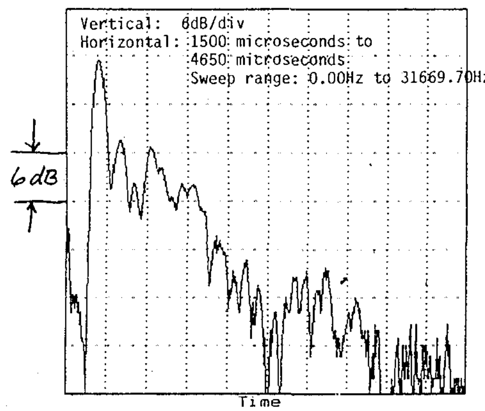


FIGURE 4. No attempt is made to align a multi-coned speaker system.

CONVERTING AN EIA SENSITIVITY RATING AND THE "ON AXIS" Q VALUE INTO AN L_W

In order to calculate L_W from a sensitivity rating L_{sensi} (i.e., *x* dB at *y* distance from Z watts) and a directivity factor Q (dimensionless), it is necessary that both ratings be obtained by means of the same general technique -- pink noise, sine wave, warble tone, etc.

When there is meaningful correlation between the two measurements (i.e., the sensitivity was read from the "on axis" Q measurement), then the following equation offers a quick way to obtain L_W:

$$L_W = (L_{sensi} \text{ at Ref. Dx and Ref. pwr.}) - (10 \text{ Log}(Q)) + \left(20 \text{ Log}\left(\frac{\text{Ref. Dx}}{0.928 \text{ ft.*}}\right)\right)$$

This equation is based on the fact that at 0.283m (i.e., 0.928 ft.), for a source with a Q = 1 (omnidirectional) the numerical value of L_W and L_p is approximately the same, within a few tenths of a dB.

$$L_W = 10 \text{ Log}\left(\frac{\text{acoustic power measured}}{10^{-12} \text{ watts}}\right)$$

Therefore, we can also easily find the acoustic power in watts once we have obtained L_W.

$$(10^{-12})\left(10^{\left(\frac{L_W}{10}\right)}\right) = \text{acoustic power in watts}$$

An interesting value to have in mind is that one acoustic watt is an L_W of 120 dB. #

WILLIAM B. SNOW & THE ENHANCED ORCHESTRA

I can't remember the exact year but it was probably between 1968 and 1970 that Bill Snow of Steinberg & Snow (Bell Labs) fame came to our house for lunch.

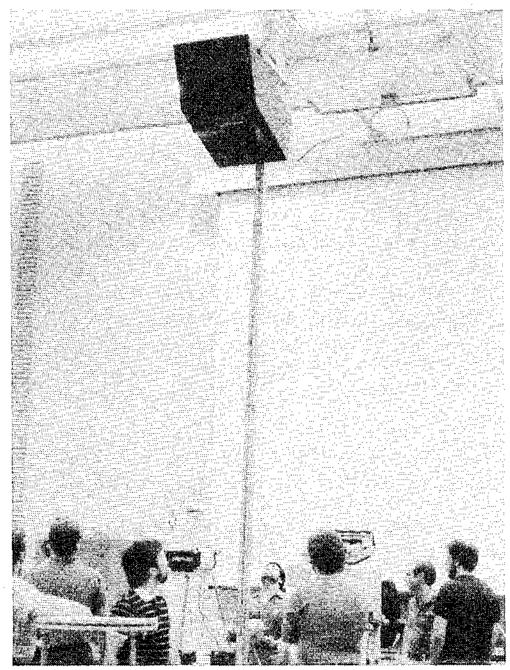
Of course, this was a thrill and we were avid listeners as he told us of his many years of research at the labs during the formative years of the development of motion picture sound.

Bill Snow's discussion of the 1933 stereophonic broadcast was the most memorable. He said that the recording and broadcast is remembered for its stereophonic aspect but what was most significant to him (and most neglected--this is late 60's remember) was that *the orchestra was enhanced by 3 dB*. Leopold Stokowski was the conductor. Stokowski wrote in his book entitled *Music For All Of Us* (Simon & Schuster, New York, 1943) "When electrical instruments are relatively perfect, they will free musicians from our present constant preoccupation with the imperfections and technical difficulties of instruments. We shall be able to give all our feeling and thought to the inner essence of the music, because the instruments will respond with extreme sensitivity to every difference of feeling in the player and the music."

The new concert hall (multi-purpose) being currently built in Munich (acoustical consultant is B.B.M.Mueller of Craemer & Mueller fame) will include Dr. Ahnert's Delta Stereophony system. One of the uses of Delta Stereophony, among other uses, will be to enhance weak instruments in the orchestra, or to quote Leopold Stokowski, ".....free musicians from our present constant preoccupation with the imperfections and technical difficulties of instruments." #

TEST OF JBL 4660 LOUDSPEAKER SYSTEM

The JBL 4660 loudspeaker system was measured at the Loudspeaker Array Design Workshop and found to be an exceptional system. In fact, whenever you can use its special coverage pattern and have the correct mounting height, your design is *successfully* done. What impressed us the most was remarkable uniformity from the front of the area covered all the way to the rear of the area covered. Shown in Figures 2a and 2b are the magnitude and phase response. Listening tests revealed that this unit sounds as good as it tested. A series of these scaled to various sizes and shapes of auditoriums would solve many a system design problem. #



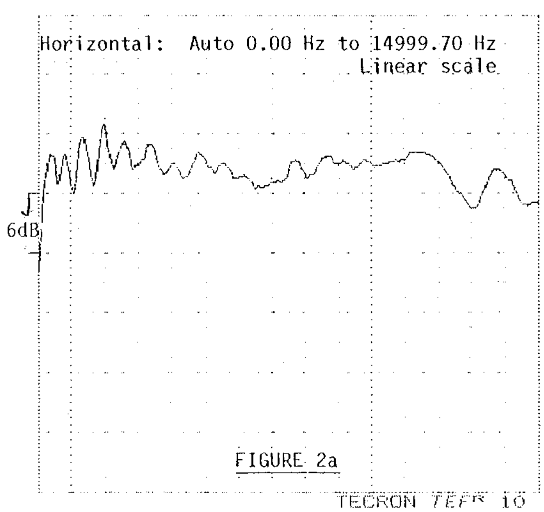


FIGURE 2a

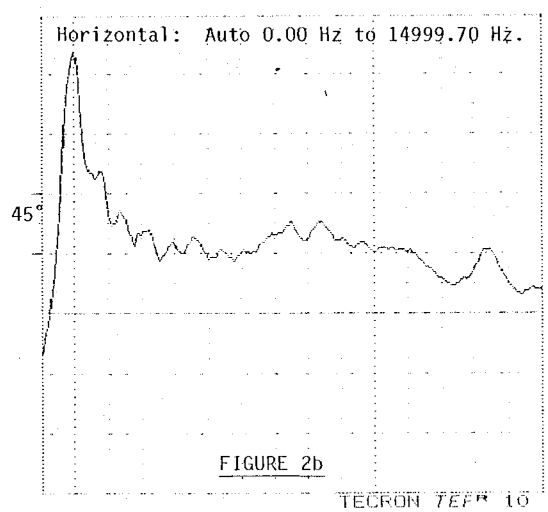


FIGURE 2b

HEYSER DISC DISPLAYS

The various versions of the Heyser disc for the TEF analyzer continues to offer access to every electrical and acoustical measurement known to mankind. Used as a digital oscilloscope (see Figure No. 1), it can also provide at the press of a few keys the FFT version of the same signal, showing us the fundamental and second harmonic (the fundamental is defined as the first harmonic) see Figure No. 2.

Heyser recently introduced in his latest disc the "Unwrap mode" wherein he corrects the phase vs frequency PFC relative origin time *in the show mode* (see Figures No. 3a and 3b. This allows the TEF user to go to the review mode, finish "tuning in" silently and then turn back on for one more test, which will be "right on the money." This is accomplished via computation and allows you to keep the new delay offset after it is computed and displayed. #

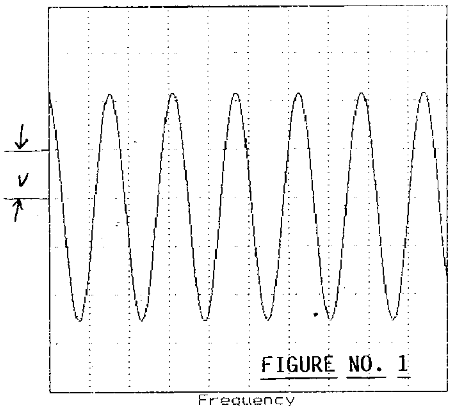


FIGURE NO. 1

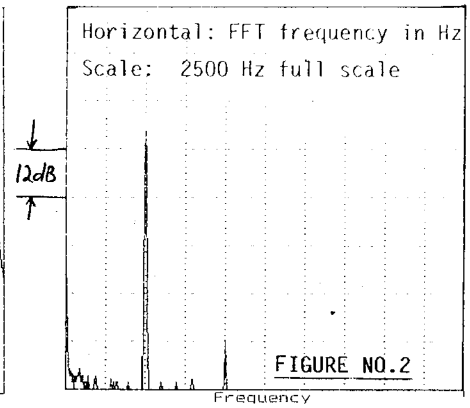


FIGURE NO. 2

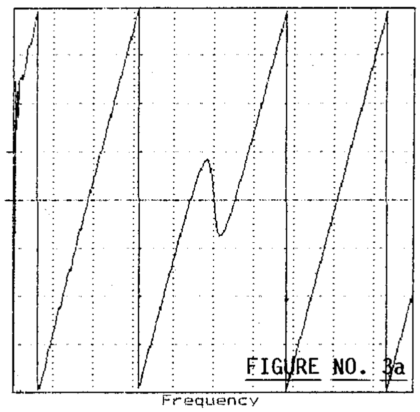


FIGURE NO. 3a

Vertical: 45 degrees/div
0 degrees is at the dashed horizontal line.
Horizontal: Auto 0.00 Hz to 19998.10 Hz
Time of test: 200 microseconds

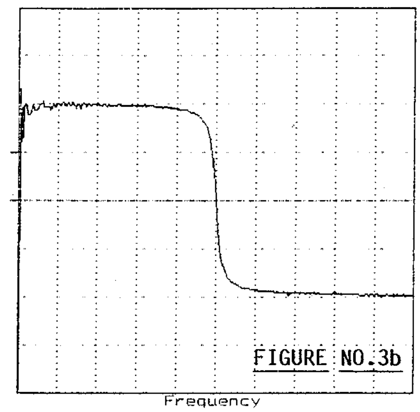


FIGURE NO. 3b

Vertical: 45 degrees/div
0 degrees is at the dashed horizontal line.
Horizontal: Auto 0.00 Hz to 19998.10 Hz
Time of Test: 0 microseconds

OLD AND NEW MICROPHONE RATINGS

The old microphone sensitivity rating of RMA was:

$$G_M = 20 \text{ LOG}\left(\frac{E_o}{P}\right) - 10 \text{ LOG } R_{MR} - 50 \text{ dB}$$

Where: G_M is the microphone's electrical available input power level expressed in dBm when driven by a sound pressure P referenced to 0.0002 dynes per square centimeter (10 dynes/CM² equals an L_p of 94 dB)

E_o is the open circuit voltage generated by a sound pressure of 10 dynes/CM²

R is the manufacturer's impedance rating for the microphone

-50dB is a constant that consists of:

-94dB (test L_p) - 6dB (open matched circuit difference) + 20 LOG (10 dynes/cm²) - 10LOG (0.001w)

Present Day Nomenclature

EIA rating (RMA) is equal to:

$$G_M = 20 \text{ LOG } E_o - 10 \text{ LOG } R_{MR} - 70 \text{ dB}$$

which is derived from: $G_M = 10 \text{ LOG}\left(\frac{(E_o)^2}{0.001\ R_s}\right) - 6.02 \text{ dB} - 94\text{dB}$

where: E_o is generated by an acoustic input of 1 pa (10 dynes/cm²).

RADIO PHYSICS COURSE 2nd Edition 1932

by

Alfred A. Ghirardi, EE

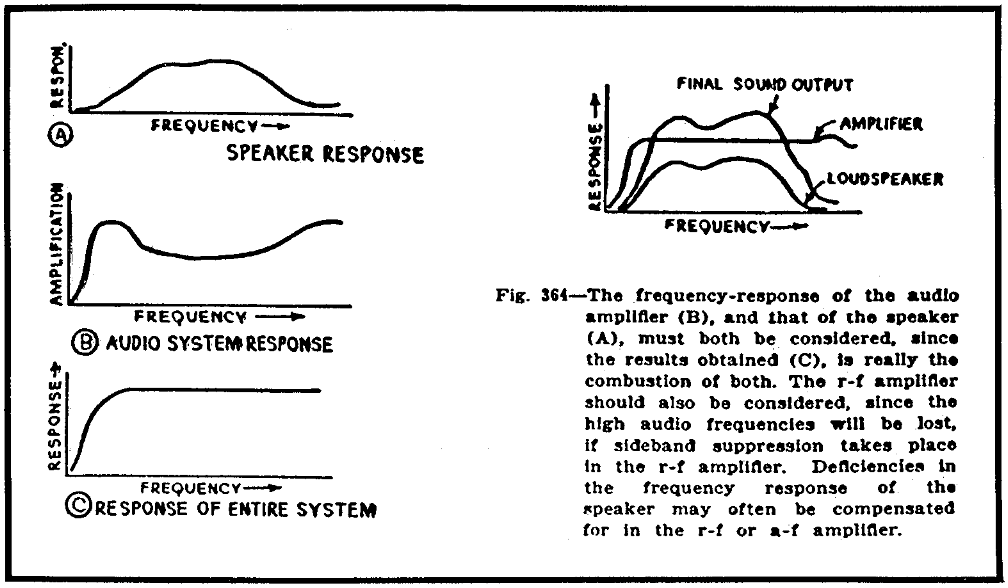


Fig. 364—The frequency-response of the audio amplifier (B), and that of the speaker (A), must both be considered, since the results obtained (C), is really the combustion of both. The r-f amplifier should also be considered, since the high audio frequencies will be lost, if sideband suppression takes place in the r-f amplifier. Deficiencies in the frequency response of the speaker may often be compensated for in the r-f or a-f amplifier.

This volume was found in the rear of a used bookstore in Gettysburg, PA. It includes pictures and details of the original Western Electric sound equipment for motion pictures marketed under the Electrical Research Products, Inc., ERPI label. It also contains what must surely be as early a reference to system equalization as it is possible to find (see figure).

The section on the decibel is far better written than most modern texts with a very clear explanation of where the multiplier 20 comes from and what the difference between voltage amplification and power gain is. #

THE VELOCITY FACTOR C_f IN TRANSMISSION LINES

The velocity of electromagnetic waves in free space is 3 x 10^8 M/S, 0.186 x 10^6 M/S, or 982 x 10^6 F/S.

The velocity c of an electromagnetic wave on a transmission line is found by:

$$c = \frac{1}{LC}$$

Where: L is the inductance in Henrys per unit of length.
C is the capacitance in Farads per unit of length.
c is the velocity of propagation.

The velocity factor c_f is found by the ratio of the velocity of the electromagnetic wave in the transmission line to the velocity of the electromagnetic wave in free space.

$$c_f = \frac{\text{c in line}}{\text{c in free space}}$$

Typical c_f for coaxial cable of the type used by broadcasters and telephone companies range from 0.66 (i.e., 66%) to 0.78 (i.e., 78%). These values for c_f allow us to find the time per mile in microseconds (usecs) by

$$\frac{1\text{ sec}}{186{,}000\text{ mi}} \cdot \frac{1{,}000{,}000\text{ usec}}{1\text{ sec}} = 5.38\text{ usec/mi}$$

in free space. For a transmission line with a c_f = 0.66, we find $5.38 \times \frac{1}{0.66} = 8.15$ usec/mi

and for a transmission line with a c_f = 0.78, we find $5.38 \times \frac{1}{0.78} = 6.9$ usec/mi

Telephone company "loaded" cable can drop significantly lower in c_f (i.e., longer in transmission time).

In the transmission of stereophonic broadcasts, the accidental routing of the two channels over routes of differing length can cause very real phase cancellation problems. Delays of from 20 to 30 usecs between channels can wipe out the entire upper spectrum. Note that it doesn't hurt for both channels to be delayed but that the delay between them is the problem.

The current Belden Master Catalog 885 lists the c_f of many cables as well as the rise time per unit of length. (Belden Electronic Wire, P. O. Box 1980, Richmond, IN 47375.) An excellent discussion of transmission lines is in the *Handbook of Electronics Calculations*, Kaufman and Seidman, McGraw Hill publishers. #

SMILE

"A Bit of Nonsense" from Newsletter Volume 12, Number 4, should read "A Bit of Fact!"

Alan Seipman, Taft Broadcasting #

SPECULAR REFLECTIONS

In large halls we use limited bandwidth frequency sweeps to obtain a large time window (T_W) ETC display.

$$T_W = (400)\left(\frac{1}{(f_2 - f_1)}\right)$$

Since D = cT   Where: D is the distance
c is the velocity
T is the time

the distance represented by the T_W is found by: $D = c\,T_W$

ETC measurements are "frequency blind." When looking at the energy returns from the surfaces of a large hall, we can therefore obtain some frequency information by dividing the bandwidth used into frequency zones. By using such "zones," we can be assured that the energy returned is at least within that frequency zone. For example, if we wished to look at a bandwidth of 2000 Hz:

Zone #1 . . . . . 0 - 2000 Hz	Zone #4 . . . . . 6000 - 8000 Hz
Zone #2 . . . . . 2000 - 4000 Hz	Zone #5 . . . . . 8000 - 10,000 Hz
Zone #3 . . . . . 4000 - 6000 Hz	

Each zone would have the same T_W and D but different frequency information.

This rough "first cut" could serve as an excellent guide to:

1. What "time areas" to investigate with increased resolution.
2. What frequency areas are providing the greatest energy returns.

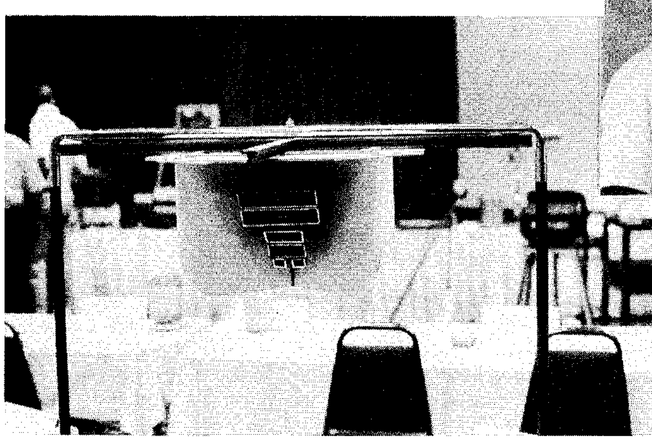
When narrow ranges of frequency are swept, remember that the "line spacing" (L_S) and the "line width" (L_W) on the ETC display are larger than those obtained using wider frequency ranges.

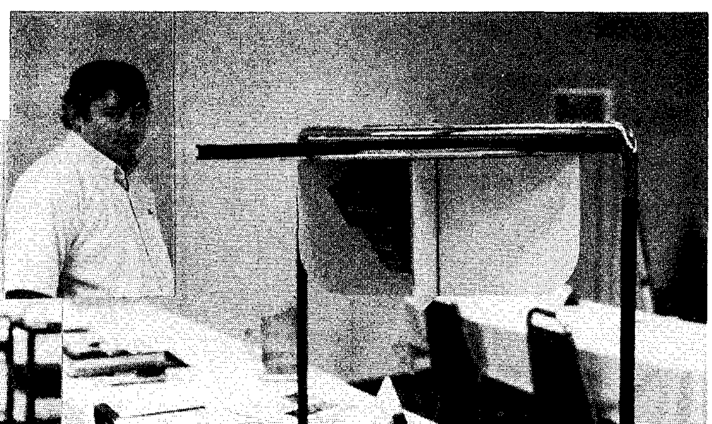
Frequency Range of Sweep	L_S	L_W	T_W
0 - 10 kHz	100 usec	136 usec	40 msec
0 - 2 kHz	500 usec	680 usec	200 msec

Use of the ETC measurements made in this fashion can lead to remarkable insights into the location of a room's critical frequency (f_C) and, subsequently, the energy returns from the diffuse field and the specular reflection field can be separately identified. With the introduction of Peutz's new "running integration" of these ETC plots, the door is wide open to the development of important new criteria for the objective judgment of acoustic environments. ♦

HOOSIER WITH AN IDEA

Wayne Staley is an energetic young Hoosier with an idea. The arrangement shown here should indeed reduce detrimental interaction at all frequencies above the barrier's wavelength.





Sonex on the sides of the "enclosure" will handle high frequency reflections. Wayne went to a lot of work to model his idea. #

CALCULATING FILTER BANDWIDTHS

There are two basic types of filter bandwidths encountered in audio devices. These are:

1. The constant bandwidth filter (CBF).
2. The constant percentage bandwidth filter (CPBF).

The constant bandwidth case is simply the assignment of an *equal* number of Hz to either side of the center frequency (f_C) thus generating a low frequency limit (f_L) and an upper frequency limit (f_U). These limits are the points at which the level has dropped 3 dB, i.e., the -3 dB frequency.

The Constant Percentage Bandwidth Filter

In the case of the (CPBF), the f_L and f_U relative to f_C varies exponentially.

The general case equation is:

$$f_{LIMIT} = b^{\pm\left(\frac{1}{2N}\right)} f_C$$

Where: f_{LIMIT} is either f_L or f_U (a plus in the exponent yields f_U and a minus yields f_L)

b is the base of the system (i.e., 2 for octave band systems and 10 for decade systems)

N is the reciprocal of the fractional octave or in the decade case 3-1/3 times the reciprocal of the fractional octave desired (i.e., a 1/3 octave filter is actually a 1/10 decade filter, thus N = 3-1/3 x 3)

Examples

What is the bandpass of one octave centered on 100 Hz?

$$f_L = 2^{-\left(\frac{1}{2 \cdot 1}\right)}(100) = 70.77 \text{ Hz}$$

$$f_U = 2^{+\left(\frac{1}{2 \cdot 1}\right)}(100) = 141.42 \text{ Hz}$$

$$141.42 - 70.77 = 70.71 \text{ Hz}$$

What is the bandpass of a 1/10th decade filter centered on 100 Hz (i.e., a 1/3 octave equalizer)?

$$N = 3\text{-}1/3 \times 3 = 10$$

$$f_L = 10^{-(1/2 \cdot 10)}(100) = 89.13 \text{ Hz}$$

$$f_U = 10^{+(1/2 \cdot 10)}(100) = 112.20 \text{ Hz}$$

$$112.20 \text{ Hz} - 89.13 \text{ Hz} = 23.08 \text{ Hz}$$

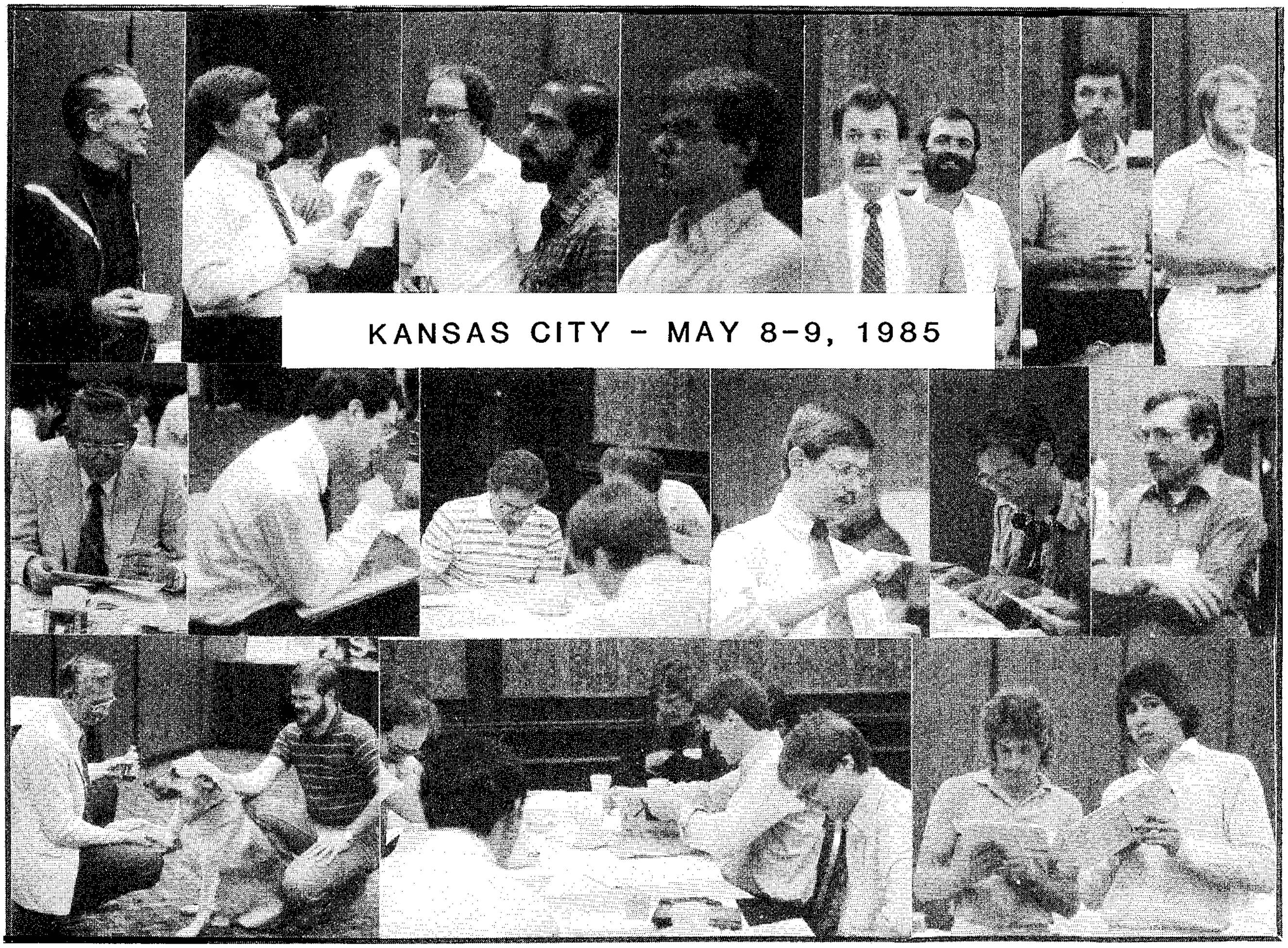
(i.e., a so-called 1/3 octave filter set with filters that are actually 1/10th decade filters has a 23% bandwidth.)

A Caution

Remember in doing these calculations that the f_C used should be the appropriate *Renard* number, not the *label* value. The advent of the modern hand held calculator has fortunately put exponential calculations within the reach of anyone willing to spend but a few minutes practicing them.

LONG - WICKERSHAM PATENT

Ed Long and Ron Wickersham have been issued a patent for a "method and apparatus for operating a loudspeaker below resonant frequency" -- United States Patent 4,481,662. For a closed box system, the drive signal is equalized by double integration resulting in a flat response below resonance and a 12 dB/octave roll off at higher frequencies. Through the use of double integration, the signal delay associated with conventional networks is substantially reduced. All you subwoofer builders should be alert to what Ed and Ron have to offer your efforts. #



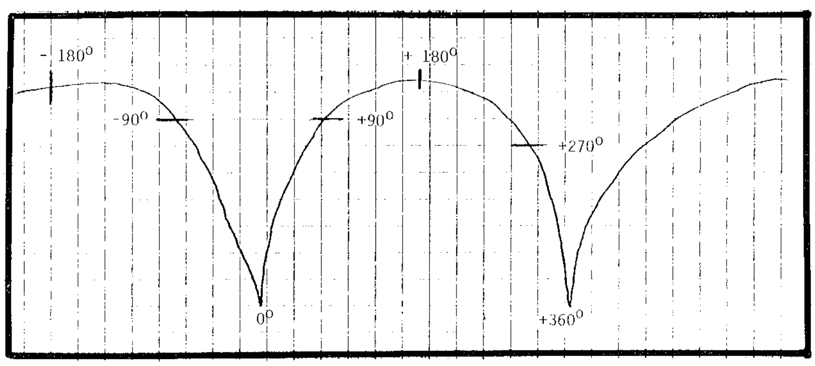
FINDING THE WIDTH OF A COMB FILTER

Useful Equations

$f_1 = \frac{1}{t}$ $\quad$ $t = \frac{1}{f_1}$

$d = tc$ $\quad$ $t = d/c$

$c = \frac{d}{t}$ $\quad$ (when $d = 1.0$ the $c = \frac{1}{t}$)



If $f_1 = 5000$ Hz

$d = (^1/_f)\ c = 0.23'$ or $2\frac{3}{4}$ "

$\frac{5000\ \mathrm{Hz}}{360^\circ} = 13.89\ \mathrm{Hz}/_{i}$

$90 \times 13.89 = 1250$ $\qquad$ $-90^\circ = 5000 - 1250 = 3750$ Hz

$5000 + 1250 = 6250$ Hz

$+180 = (180 \times 13.89) + 5000 = 2500 + 5000 = 7500$ Hz

$+360 = (360 \times 13.89) - 5000 = 5000 + 5000 = 10{,}000$ Hz

Therefore, the -3 dB bandwidth is equal to: 6250
-3750
2500 Hz

THE SYN-AUD-CON MANAGEMENT FORUM

A worthy addition to the literature of "Murphy's Law" is this study in depth of projects and their mismanagement. Many texts are available on the subject of management and academia has established elaborate management courses. What's astounding in all this is that, like so many things we are taught in school, real life is just not like that. Management is easy to cope with but what a majority of us face in real life is having to live daily with mismanagement. Learning to understand the cardinal principles of mismanagement leads to an understanding of its basic nature and can lead in extreme cases to becoming the mismanager of your own business. It's in this spirit that we present the following fundamentals:

The Seven Phases of a Project

1. Wild enthusiasm
2. Total confusion
3. Disillusionment
4. The search for the guilty
5. The punishment of the innocent
6. The departure of the competent
7. The promotion of the non-participants

The steps above are those most often encountered in free enterprise, private, profit seeking ventures. When the enterprise is a monopoly or a governmental one wherein profit elicits a sneer, then the following rules seem to apply:

The Three Classes of Employees in Non-Profit Organizations

1. The F.O.F.'s - (One translation is "Faithful Old Followers")
2. The Golden Boys - (Those selected at the outset to rise to the top by having met whatever "Good Old Boys" criteria are present)
3. PO-PO's - (The "poo-poos" are the pee'd on and passed over group)

One remarkably insightful rule that applies universally to all groups is:

"The boss may not always be right, but he's always the boss."

and expresses succinctly the reality of our society. One of the most illuminating experiences there is results from a close association with employees while you're their boss followed by further close association after you've resigned and taken up other work. The filter in the feedback path to the boss is a remarkable mental mirror of what the boss projects carefully adjusted to entertain, not inform. #

THE PRESSURE ZONE IS FREQUENCY DEPENDENT

Referring to the table on dB addition in original LEDE Tech Topic V8 N2, in figure 1 we find that 60^o is with 1.0 dB of coherent addition of two signals. One wavelength of 40 Hz is:

$$\lambda = \frac{1130}{40} = 28.25'$$

$$\text{and } \frac{360^o}{60^o} = 6$$

Therefore, $\frac{28.25'}{6} = 4.71'$ as a logical depth of the *pressure zone* for 40 Hz. Using the same reasoning we would find:

$$x = \left(\left(\frac{1130}{1000}\right)/6\right)12 = 2.26"$$

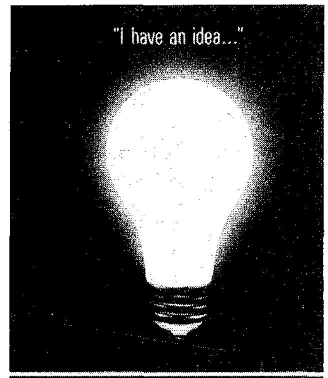
$$x = \left(\left(\frac{1130}{10000}\right)/6\right)12 = 0.23"$$

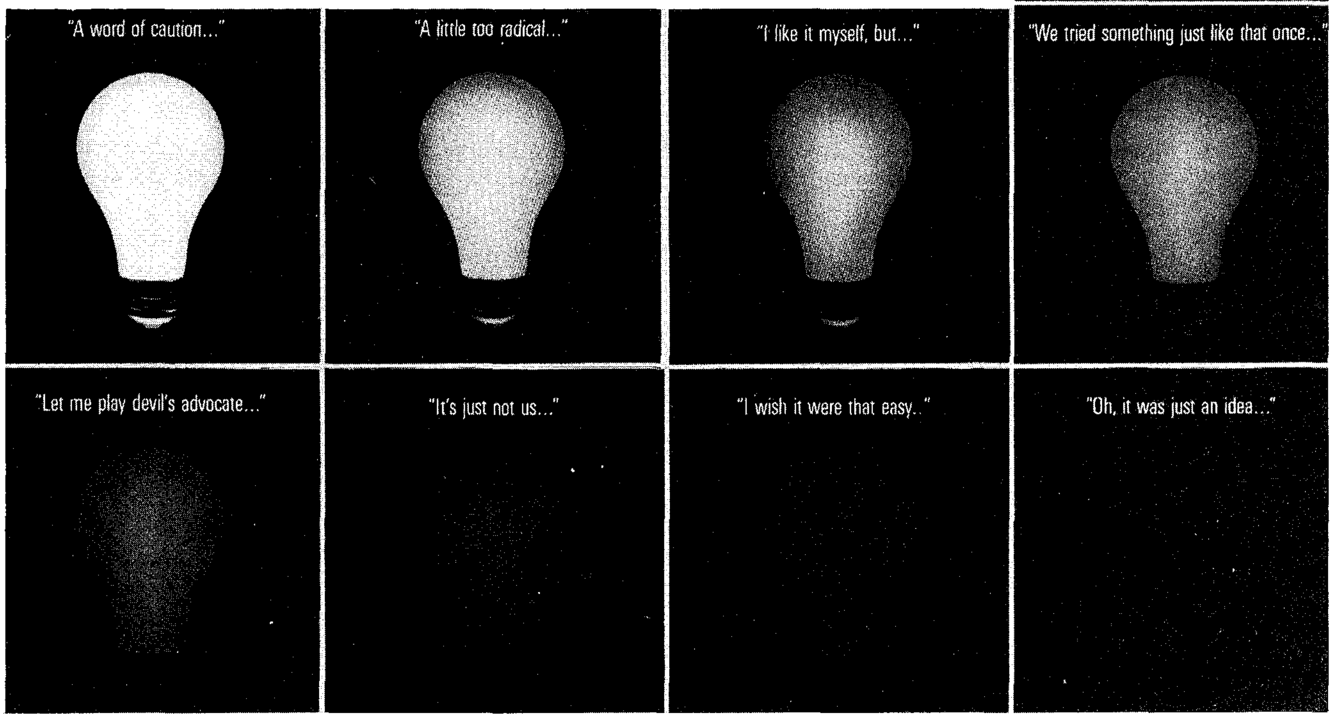
$$x = \left(\left(\frac{1130}{20000}\right)/6\right)12 = 0.11"$$

As we have indicated previously, the pressure zone can be effectively used for subwoofer response in small rooms with sufficiently rigid boundaries - basements or in automobile interiors.

TO KILL AN IDEA

Ideas are universal. The "why" behind American industries' problems with competition from abroad is enshrined in this illustration. #





TYPICAL AVERAGING TIMES FOR 1/3 OCTAVE ANALYSIS

If it is desired to obtain a 1/3 octave measurement of similar accuracy with respect to level as that obtained by TEF® analysis, Brüel and Kjaer suggests around 500 samples (N_S) as a realistic statistical base. Bandwidth (B) and averaging time (Ta) are related to this sampling in the following manner:

$$N_S = B(Ta)$$

From this equation, it can be seen that to have an accurate estimate of a 1/3 octave band's level at 100 Hz (bandwidth 23 Hz) would require

$$Ta = \frac{N_S}{B} = \left(\frac{500}{23}\right) = 21.7 \text{ secs}$$

While for 1000 Hz (B = 230 Hz), the averaging time required would reduce to:

$$\left(\frac{500}{230}\right) = 2.2 \text{ secs}$$

And finally for 10,000 Hz (B = 2300 Hz), we find

$$\left(\frac{500}{2300}\right) = 0.22 \text{ secs}$$

of averaging time required.

This is, of course, why those low frequency markers jump around so randomly on your 1/3 octave real time analyzers with the one to two second integration times while the high frequencies seem relatively stable.

While there are those who think they are measuring low frequencies with such units, they can learn better with experience. ♦

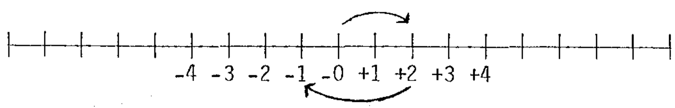
MATHEMATICAL OPERATIONS

Add

Adding is taking the sum of two numbers (1 + 1 = 2) or counting the total number.

Subtract

Subtracting is changing the direction of addition from 0^{o} to 180^{o}.



+2 -3

means do two marks in the positive *direction* from zero followed by three marks in the *negative direction* from the last positive mark you were at (+2). This results in a final position on the marks of -1.

Multiply

Multiplication is a form of repeated addition.

$$(3 \times 6),\ (3 \cdot 6),\ (3)(6)$$

All mean add three sixes together: 6 + 6 + 6 = 18.

Divide

Dividing is a form of repeated subtraction.

$$(6 \div 3),\ (6/3),\ 3\overline{)6}$$

All mean find the number that when subtracted from 6 3 times results in zero.

$$+6 -2 -2 -2 = 0$$

Powers

Taking numbers to higher powers (exponents) is a form of repeated multiplication.

$2^{(4)}$, 2 exp 4

All mean multiply 2 by itself 4 times.

Multiplication No.	1	2	3	4
Multiplication	2 x 2 x 2 x 2 = 16			

Roots

Roots are a form of repeated division.

$$16^{(1/4)},\ \sqrt[4]{16},\ 16 \text{ exp } 0.25$$

All mean find the number that can be divided into 16 four times and results in zero.

Logarithms

Taking the logarithm of a number is a method of expressing that number as an exponent of some chosen base number.

Using the base 10 allows simple illustrations to be formed. $10 \times 10 = 100 = 10^2$

$$\underset{\text{base}}{\uparrow}\!\!\!\!\!\!\mathrm{LOG}_{10}\quad \underset{\text{number}}{100} \quad = \quad \underset{\text{logarithm}}{2}$$

Logarithm operation means take the LOG of the number to the base.

Antilog

The inverse of this operation is the number ratio expressed as an exponent of the base.

$$\mathrm{LOG}^{-1},\ \text{antilog, all mean } 100/1 = \underset{\substack{\uparrow \\ \text{antilog}}}{10^{(2)}}$$

Continued on next page...

Log Multipliers

In audio and acoustics we use logarithmic multipliers such that:

$$M\ LOG_b\ a/c = NM$$

Where: M is the multiplier

$LOG_b(a/c)^M$ is identical to $M\ LOG_b\ a/c$

b is the base (may be any value other than zero or unity)

a/c is the ratio being converted into a logarithm

NM is the logarithm times the multiplier

Antilogs of Multiplied Logarithms

Step number one is to remove the multiplier so the quantity can be treated as a normal logarithm. The inverse operation, of course, is division so that:

$$\frac{NM}{M} = N \text{ is obtained}$$

Then the antilog is found by the standard method: $b^N \doteq a/c$

These two valuable tools are written as: $M\ LOG_b\ a/c = NM$ (log form)

$$a/c = b^{\left(\frac{NM}{M}\right)} \text{ (antilog form)}$$

Complex Numbers

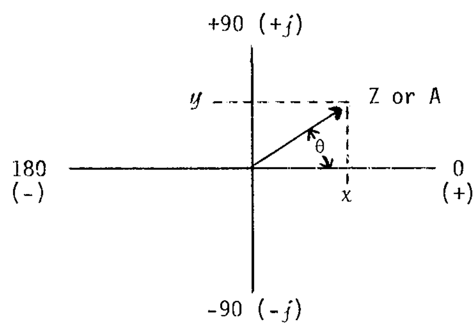
Complex numbers are numbers with more *directional information* than conventional numbers. Complex numbers can be written as:

$x \pm jy$ (rectangular form)

or as: Z or $A\ e^{i\theta}$ } polar form

or as: Z or $A\ \angle\theta$ }

Just as in the case of the plus symbol (an instruction to go toward zero degrees) and the minus symbol (an instruction to go towards 180°), the j, i or $\sqrt{-1}$ symbol instructs you to go up to 90° ($+j$) or -90° ($-j$) on the argand diagram.



The diagonal formed inside the rectangle outlined by $x + jy$ is called a "vector" and is used in audio and acoustics for impedance (Z), amplitude (A), and other complex quantities. The symbol θ indicates the angle between the 0° axis and the vector. θ can be expressed in degrees or radians and when radians are used, the notation:

$$e^{i\theta}$$

is commonly employed.

$$e^{i\theta} \equiv \angle\theta \qquad \theta = \arctan\frac{y}{x}$$

and

$$x = Z \text{ or } A \cos\theta$$

$$y = Z \text{ or } A \sin\theta$$

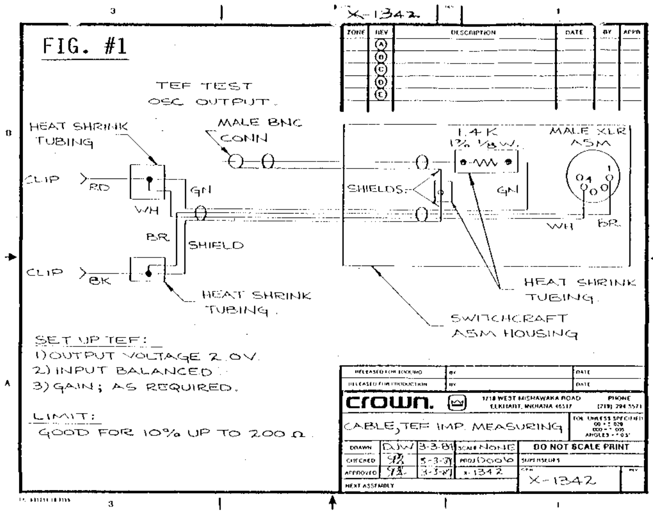
LOUDSPEAKER IMPEDANCE MEASUREMENTS

Tecron demonstrated its special impedance measuring cable (see Fig. #1) (measures accurately up to 200Ω) 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 1 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



Example Measurements

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.

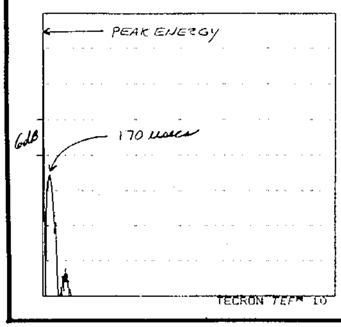


FIG. #2

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

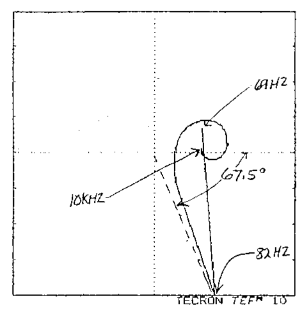
Horizontal: 0 microseconds or 0 FT to
6278 microseconds or 7.09364 FT
scale: 1.9395E+00 FT/inch or 7.6357E-01 FT/cm.
1716 microseconds/inch or 675 microseconds/cm.

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 31781.00H

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

The Nyquist phase plot (NPP) of the resistive and reactive components of the impedance shows a low frequency capacitive reactance phase angle of approximately -67.5° (see Fig. #3). 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 versus frequency curve (PFC).

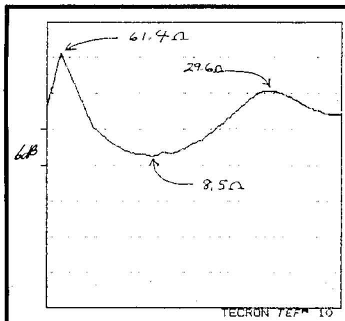


FIG. #4

FIG. #5

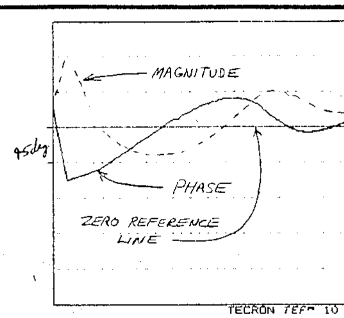
Parameters for Fig. #4 and Fig. #5:

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



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 the wider range at this rate, each sweep would have taken

$$\frac{(10{,}000 - 70)}{5 \cdot 60} = 33.1 \text{ minutes}$$

As it was, the sweep took:

$$\frac{248 - 70}{5} = 35.6 \text{ secs}$$

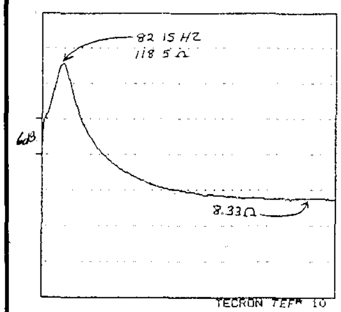
Continued next page...

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

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

Our earlier measurement had a $T_R = \dfrac{14 \text{ Hz}}{200 \text{ Hz/sec}} = 70$ msec

FIG. #6



Vertical: 6dB/div with base of display at 39.9dB
0dB 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

Vertical: 6dB/div with base of display at 39.9dB 6dB
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

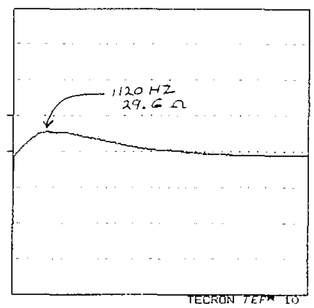


Fig. #7 is the high frequency resonant peak done at a slower sweep and higher frequency resolution. Note here that the high frequency resolution was sufficient in the earlier measurement. These last two figures both have *linear* frequency scales whereas the overview figures are logarithmic frequency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

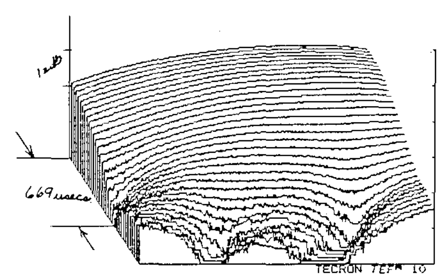


FIG. #8

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_w = 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. #

CLASSIFIED

FOR SALE: Altec Master Test Set which includes 1/3 octave filters, 1 octave set, meter attenuator, hi & lo pass filter set. Also other 1/3 octave filter set. Make offer.

CONTACT: Mike Sullivan, Thunder & Lighting (713) 529-8402.

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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.

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