SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

newsletter

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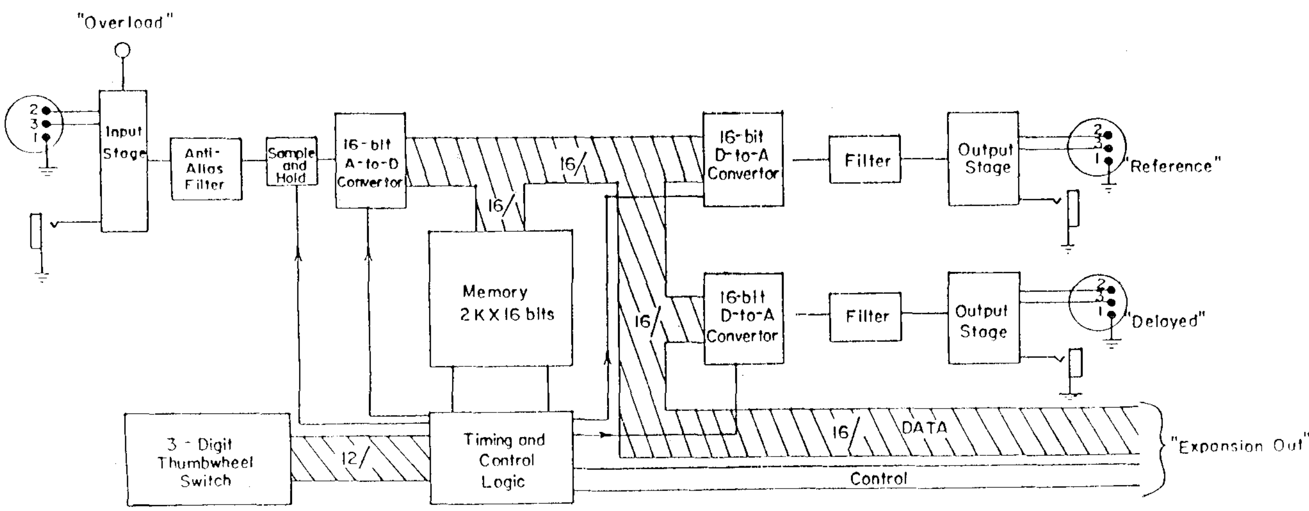
SYNERGETIC
Working together; co-operating, co-operative

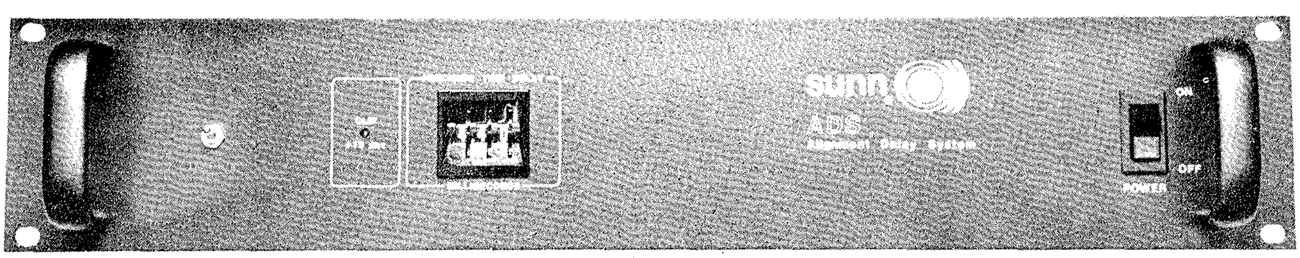
SYNERGISM
Co-operative action of discrete agencies such that the total effect is greater than the sum of the two effects taken independently.

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

SUNN DOES IT!





10 MICROSECOND/STEP SIGNAL ALIGNMENT

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SYN-AUD-CON SCHEDULE - FALL/WINTER 1983

Starting in September we will be conducting two-day classes in the following cities. These classes run from 8:30 A.M. until 5:00 P.M. Lunches are provided as well as morning and afternoon coffee breaks. The price is $395.00.

It is our sincere belief that you will find these classes well worth attending because they are different from any previous classes we have done.

2-DAY SOUND ENGINEERING SEMINARS

Sept. 13-14	St. Louis, MO	Nov. 7-8	Orlando, FL
Sept. 29-30	Chicago, IL	Nov. 15-16	Dallas, TX
Oct. 6-7	New York, NY	Nov. 29-30	Houston, TX
Oct. 18-19	Washington, DC	Dec. 13-14	Las Vegas, NV
Oct. 26-27	Atlanta, GA	Jan. 18-19, 1984	Anaheim, CA

OCTOBER - Atlanta, GA

Tentative TEF Instrumentation Workshop With Dr. Eugene Patronis

The things we no longer do in these classes that have allowed us to shorten dramatically the time required is to eliminate instruction on the HP-41C and drop a majority of our product type demonstrations. We have completely rewritten our Syn-Aud-Con Lab Manual to allow detailed discussion of the basics all of us should know like the back of our hand.

We have now used our Tecron TEF® analyzer since last November. We have so many exciting discoveries about devices you regularly use in your systems that we can hardly wait to share them with you. You'll find it hard to believe the speed and accuracy with which we now make measurements that a year ago did not exist, such as 3-D directivity plots of loudspeakers and microphones. This newsletter shares the *beginnings* of this new data. We will be *demonstrating* in these classes the real time signal "line up" of multiple loudspeakers by using the relative phase angle vs frequency measurement on the TEF® analyzer while doing the actual adjustment on the spectacular Sunn microsecond signal delay device.

We sincerely believe that one who has not witnessed for himself or herself the power of these new analytic tools will have sacrificed one of the real advantages of being part of Syn-Aud-Con -- that of seeing *way* in advance what's going to happen in your industry.

You'll like the shorter hours, lower price, and the same high quality of shared ideas as always.

MICROPHONE WORKSHOP IN PARIS, 1984?

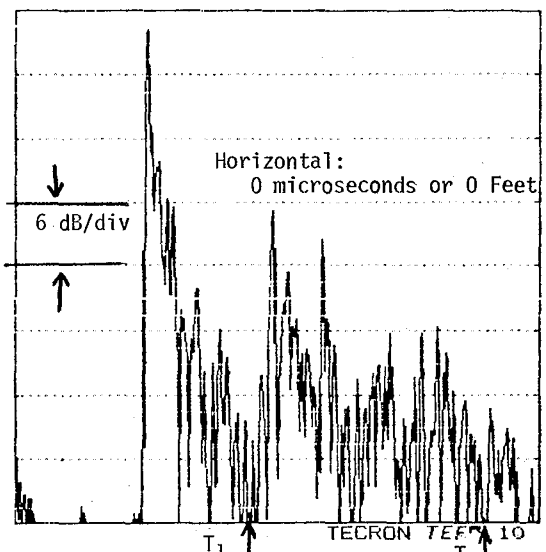
Syn-Aud-Con has organized some exceptional workshops during this past year: TEF® Workshop, Loudspeaker Array Design Workshop, Concert Hall Design Workshop. These workshops have had truly outstanding, proven experts head them up. Dr. Patronis, Dave Klepper, V.M.A. Peutz, and others of this caliber have formed the staffs.

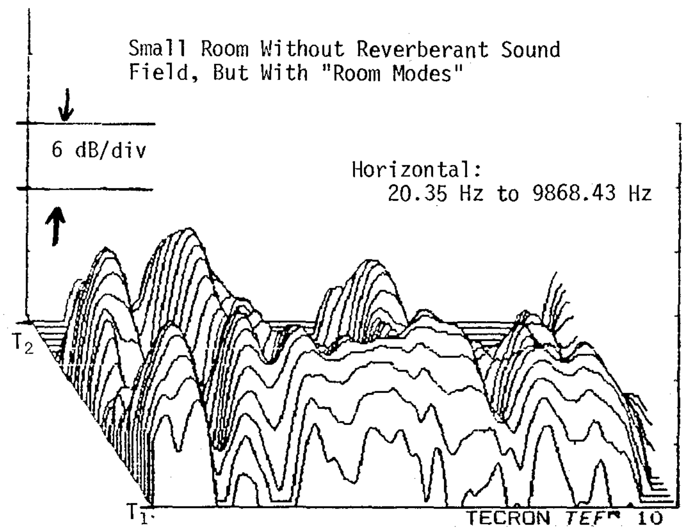
After the success of our Concert Hall workshop in Europe, we are considering the possibility of a "Microphone Workshop" in Europe. We'd like to know how many of you would be interested in a 3-4 day microphone workshop dealing with how to choose and use contemporary microphones in recording classical orchestras, operas, and contemporary groups, with a visit to a concert hall, opera house, and bistro to do just that. Our staff would consist of European experts in each of these fields from manufacturing firms to independent recording engineers. Price would run at $1,000 - $2,000 per person, depending on the scope of the program and staff. We would include lunches, dinners, and transportation to recording sites. Each participant would pay his own way to and from Europe (probably just before or after AES in Paris), and his own hotel expenses, though we would locate and reserve hotel space for the group.

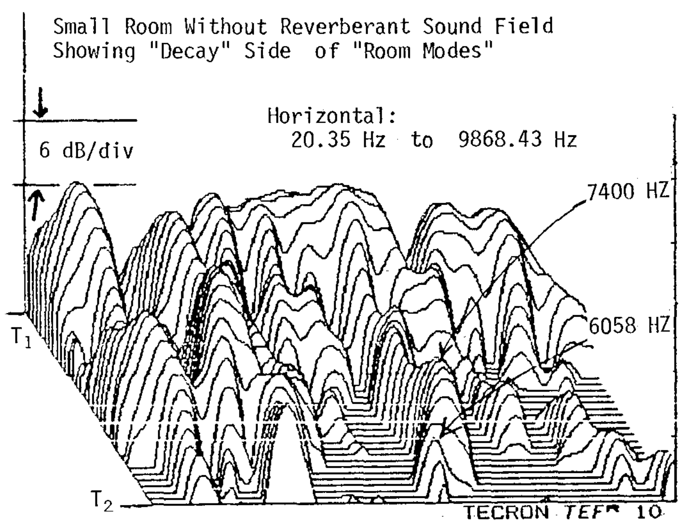
Let's hear from you if you are interested.

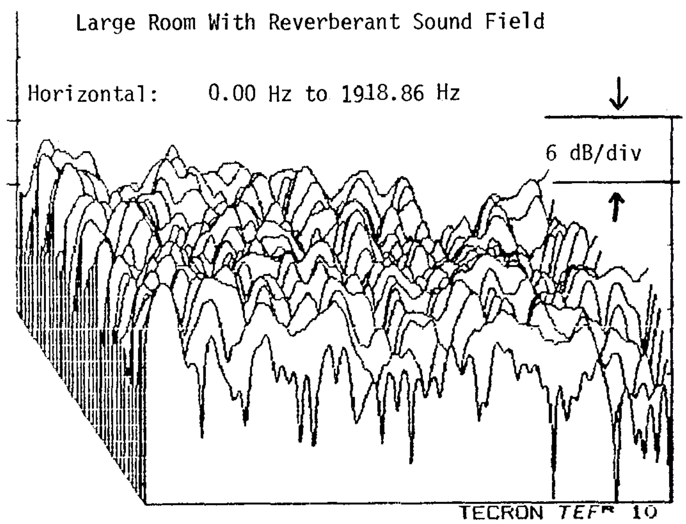
ROOM MODES

Here is vivid proof that there is a fundamental difference between a small reverberant space and a large reverberant space.







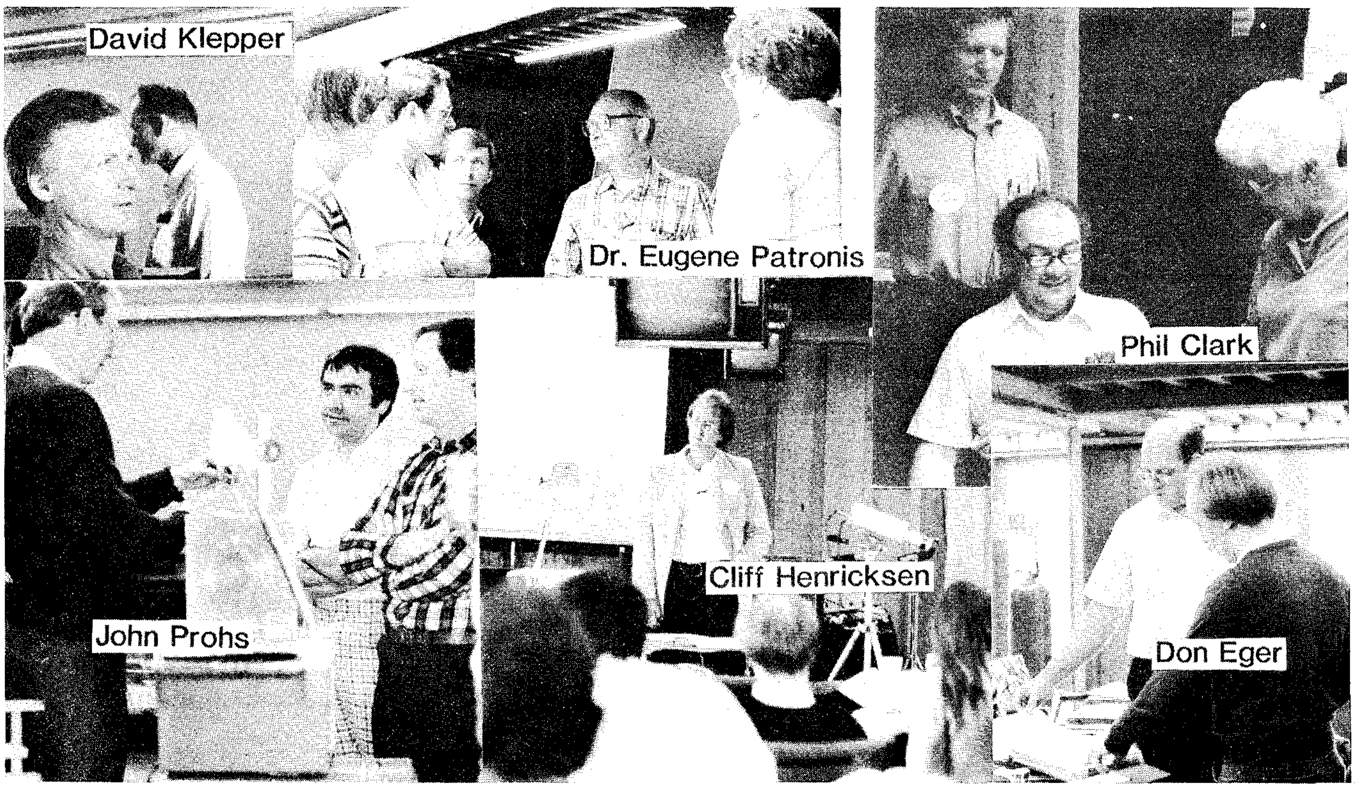


PZM® PATENT ISSUES

United States Patent number 4,361,736, issued November 30, 1982, to ED LONG and RONALD J. WICKERSHAM: Five key claims were allowed, including its use as an acoustic filter. The patent is entitled "Pressure Recording Process and Device." Thus, both the pressure zone process of recording and the pressure zone microphone are protected. PZM® is a registered trade mark and the "plate type mounting" constitutes "Trade dress." Infringers, therefore, need to avoid similar names, appearances, and performance claims if they wish to avoid potential legal entanglements.

APRIL LOUDSPEAKER ARRAY WORKSHOP

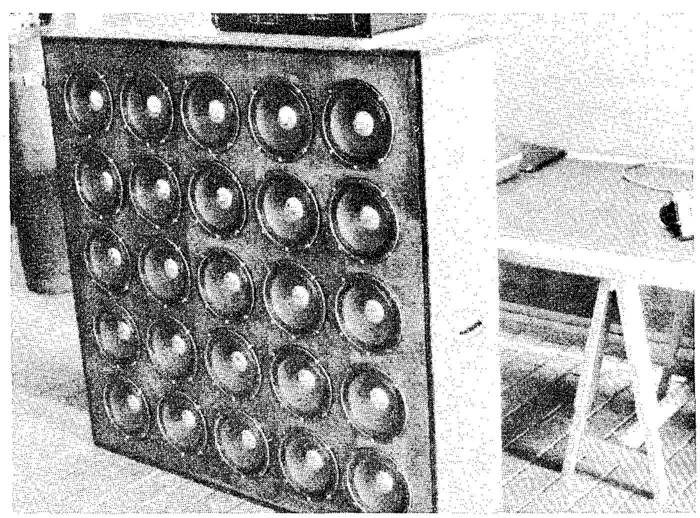
What makes a great workshop?........Superb staff and a class made up of manufacturers, consultants and highly involved and motivated designers of sound systems.


David Klepper
Dr. Eugene Patronis
Phil Clark
John Prohs
Cliff Henricksen
Don Eger

Rethinking Arrays of Loudspeakers

What's the difference between *a loudspeaker array* and *an array of loudspeakers*? A loudspeaker array consists of devices differing in sensitivity, Q, electrical power input, etc., assembled into an array intended to act as a single source of sound. An array of loudspeakers is a mass *of the same kind of device* arrayed in one area, i.e., a ceiling.

For as long as this writer can remember, very experienced sound men cautioned against series-parallel connections of loudspeakers in a large ceiling system. Strange effects had been noted over the years, usually attributed to unintentional networks being formed, when a large number of identical loudspeakers were series-parallel connected rather than in parallel across a constant voltage line. It changes the directivity of the array.



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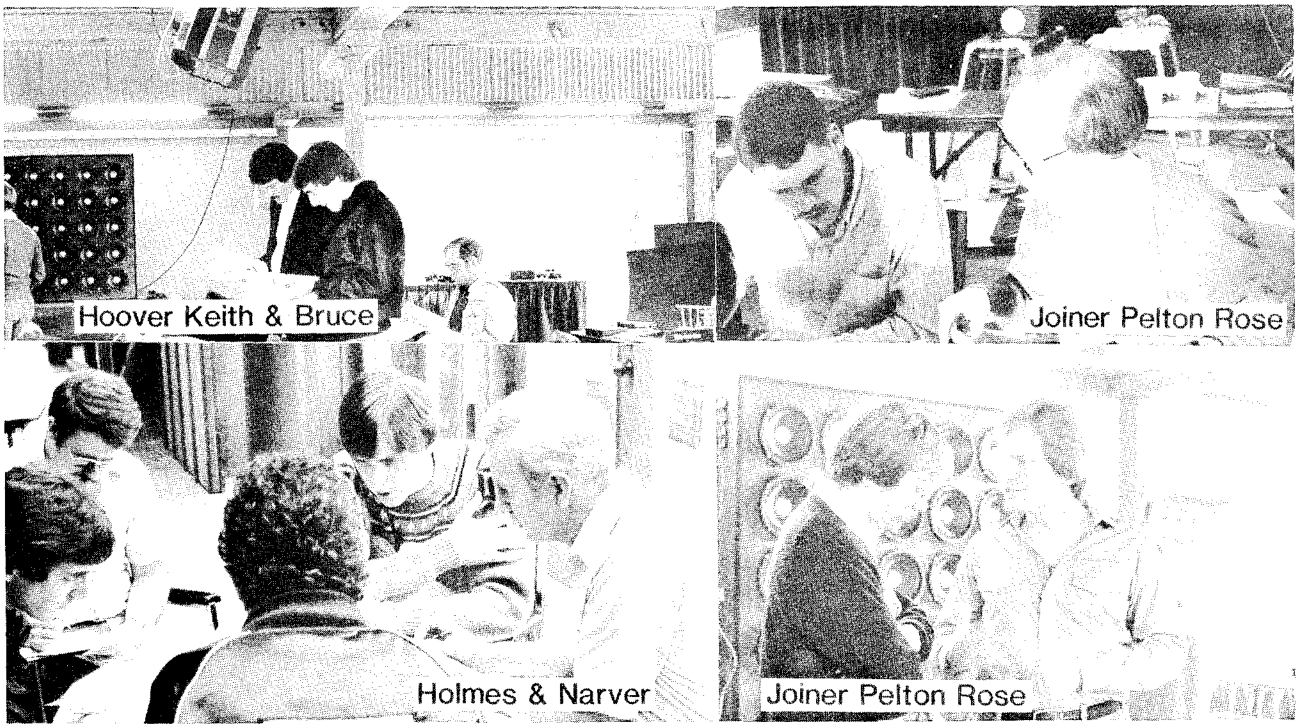
A Breakthrough

Philips, Eindhoven, demonstrated a new technique for handling arrays of loudspeakers at the AES Convention in their city this spring. For our Loudspeaker Array Workshop, Manny Mohageri of Emilar built an array of 25 identical 8" loudspeakers and KEN WAHRENBROCK wired a relay system that allowed us to switch between all 25



in an Nth order Bessel function. The switch is startling to say the least. It goes from a lobing mutually destructive interference pattern to a mutually constructive pattern that has essentially the polar response of a single 8" speaker while having the power handling of 25 speakers. Outdoors that translates into about a 160°

horizontal coverage through the articulation region. A crosswind of 15-20 MPH simply didn't affect it as the pattern was "lensed" about; you remained in it because of its wide angle and, therefore, didn't hear the usual level differences.

Continued next page.....

We'll have much more to say on this later. Let me assure you, the combination of Don with a Tecron TEF® analyzer, Gene Patronis with a thorough understanding of Bessel functions, and Manny Mohageri and Ken Wahrenbrock to build and wire them is, at present, a mandatory requirement for a successful application of the principle.

Demonstrations were conducted where observers first heard a single 8" unit in its own enclosure, followed by all 25 in the special connection, followed finally by all 25 hooked in parallel. To walk across the horizontal coverage for each of these cases is an ear opening experience.

Now, imagine 25 super horn speakers of 25 times the power but with the coverage pattern you desire rather than the lobing usually present and you'll see that these loudspeaker array classes may become arrays of loudspeaker classes.

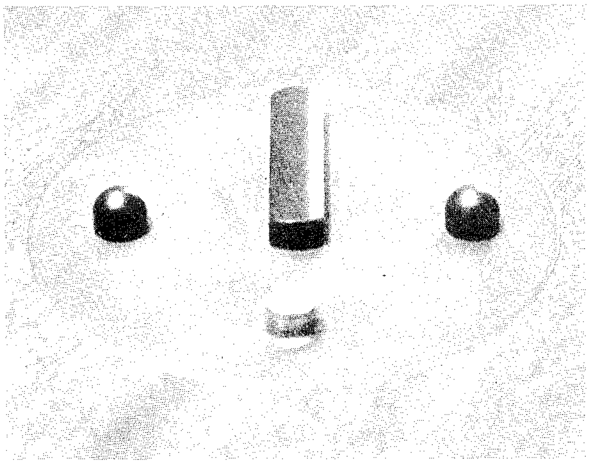
SMILE

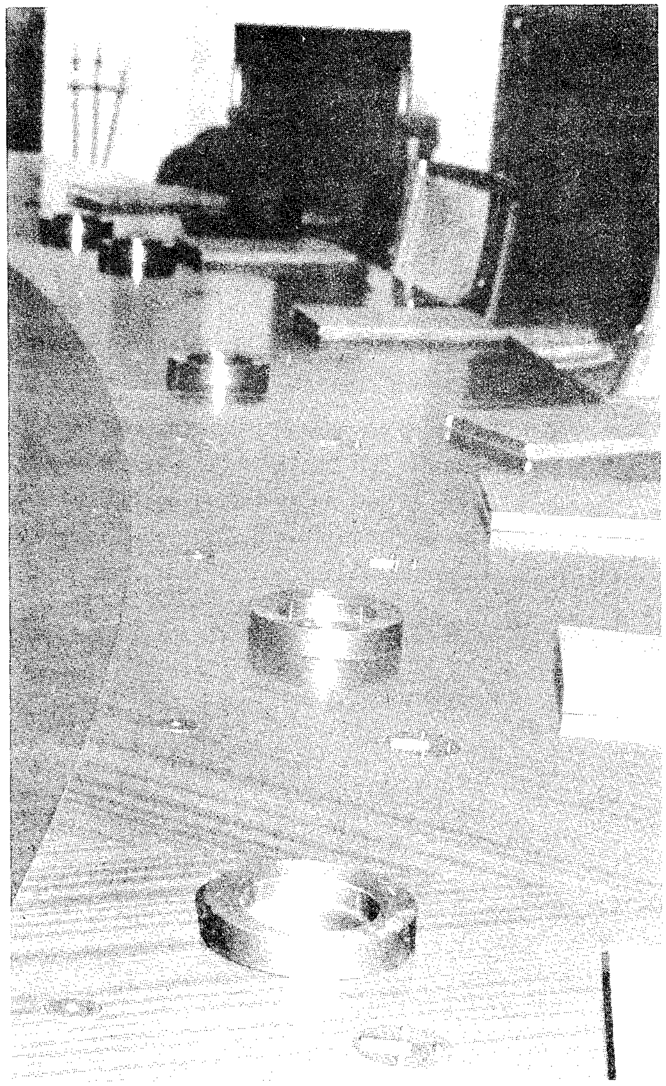
A French balloonist drifted off course and across the channel. He landed in a field in England. A man came running across the field. The Frenchman asked him where he was. The Englishman said, "You are in a basket in the middle of a field." The Frenchman asked if he was an accountant. The Englishman said, "Yes, how did you know?" The Frenchman said, "What you said was absolutely accurate but totally useless."

PZM's® AT THE FEDERAL RESERVE

PAUL FREEMAN, Supervisor at the Board of Governors, Federal Reserve System in D.C., sent in the following:

> *"I consider* (this system) *to be one of the most deluxe PZM® installations in the free world.*
>
> *They began life as PZM 20 RMG's. The plates were machined out of ¼" brass (± 2 mils!) and have the power supplies remoted under the table.*
>
> *There are 20 microphones total on the table (20" in from the edge and approximately 30" apart), which interface to an Ivie automatic mixer. An automatic mixer is vital in using a multi PZM configuration.*
>
> *Needless to say, the powers that be here at the Fed are tickled to death with the appearance and clarity of the system.*

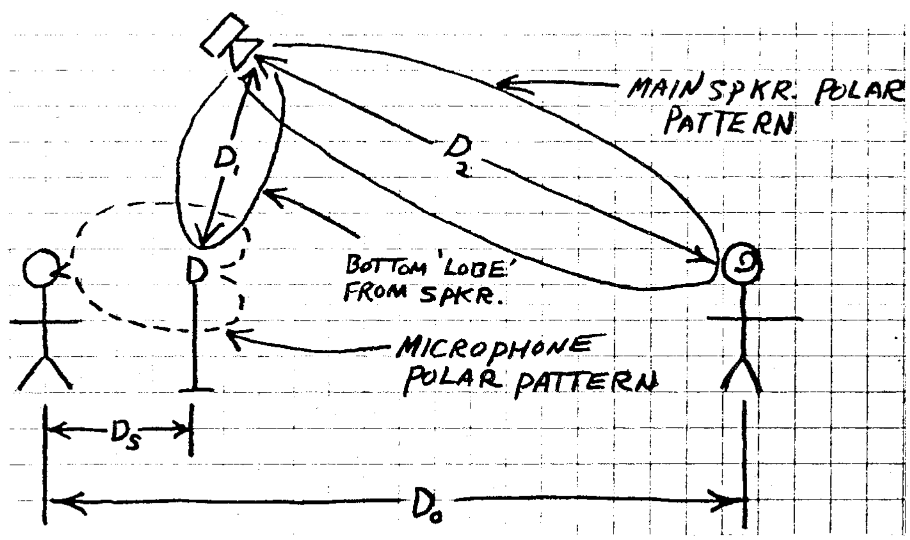




PZMs at the Federal Reserve Board, Wash. D.C.
Power supplies remoted under the table.

SOUND RADIATION PATTERNS RELATIVE TO THE ELECTROACOUSTIC MODIFIER (Me)

1. Take the difference , in dB, between the level on the mic. polar plot at the angle toward the talker and the angle toward the loudspeaker (MS_L).
2. Take the difference, in dB, between the level on the loudspeaker's polar plot at the angle towards the listener and the angle toward the microphone (SM_L).
3. $M_e = 10 \text{ EXP } (MS_L + SM_L)$
4. $PAG_{(OMNI)} + (MS_L + SM_L) =$ Total Gain (free field)

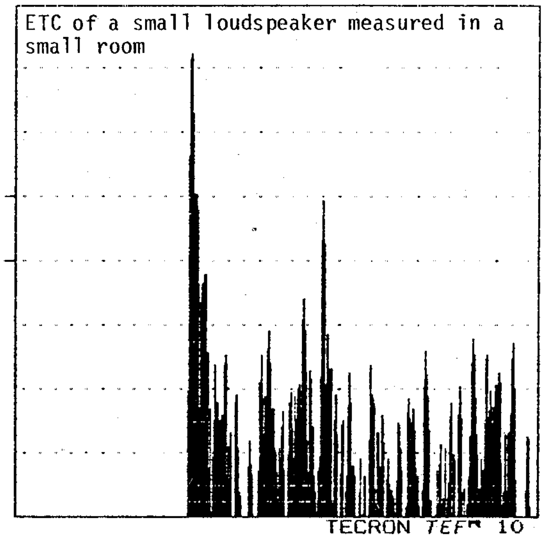


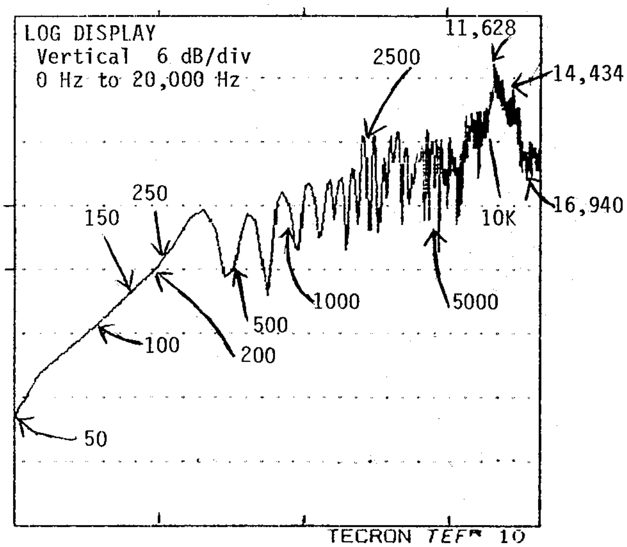
TEF® ANALYZER ADDS LOGARITHMIC DISPLAY

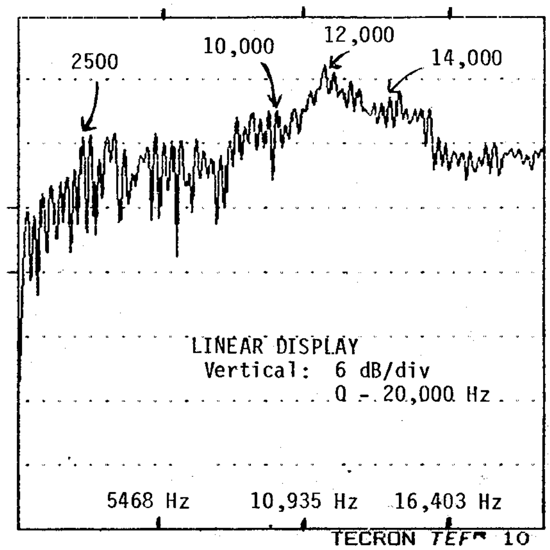
Right at "press time" for this Newsletter, we received a new disc (VER 1.0E) from Tecron for our TEF® analyzer. It is spectacular!

We are truly finding it hard to believe the improvements we are witnessing in this technology. Among too many major advances to catalog here is the ability to have the energy frequency curve, EFC, phase vs frequency curve, PFC, and Nyquist phase curve, NPP, plotted on a logarithmic frequency scaling. Dick Heyser has reported to us that the TEF® analyzer and his personal equipment correlate to fractions of a percent.

Gerald Stanley's responses to requests for rapid access to all measurements has exceeded our wildest expectations and the special "tools" he is including in the software is exceptional in every imaginable way.







EFFECT OF DIRECTIONAL DEVICES ON ACOUSTIC GAIN

I. Find PAG for an omnidirectional source ($PAG_{(OMNI)}$).

II. Find *relative** change in level (L_X) for:

(A) Microphone to talker angle (M→T)

(B) Microphone to loudspeaker angle (M→L/S)

(C) Loudspeaker to listener angle (L/S→ℓ)

(D) Loudspeaker to listener angle (L/S→M)

*Relative normally to "on axis"

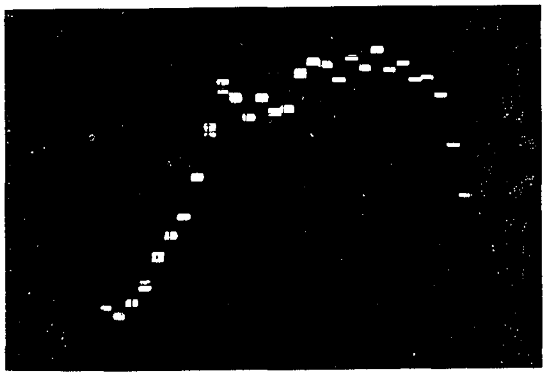
III. Use equation below:

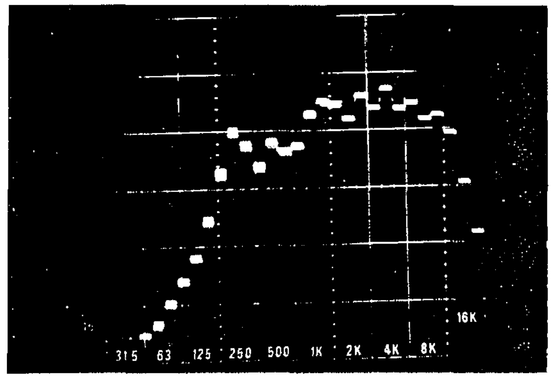
$$PAG_{(OMNI)} + (L_{(M\to T)} - L_{(M\to L/S)} + L_{(L/S\to \ell)} - L_{(L/S\to M)})$$

$$M_e = 10^{\left(\frac{\text{Above brackets}}{10}\right)}$$

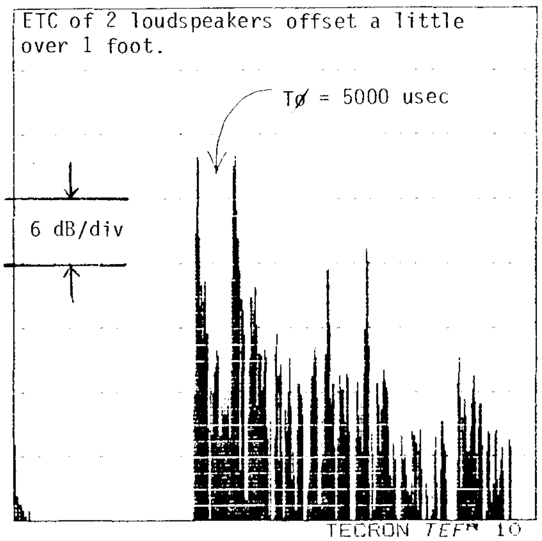
LOG/LIN VIEW OF COMB FILTERS

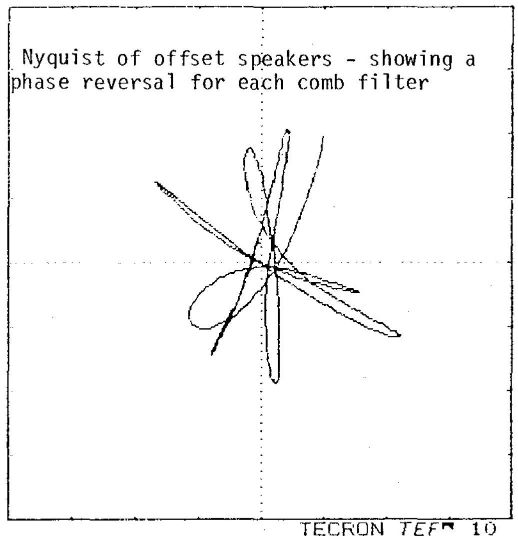
We were the first in audio to use real time analyzers (in 1968) in conjunction with 1/3 octave equalization. I believe this photograph of a real time analyzer will announce the recognition of the limitations of 1/3 octave RTA's. It may take 10 years, just as it took 10 years to put a real time analyzer in the hands of 90% of the sound contractors.

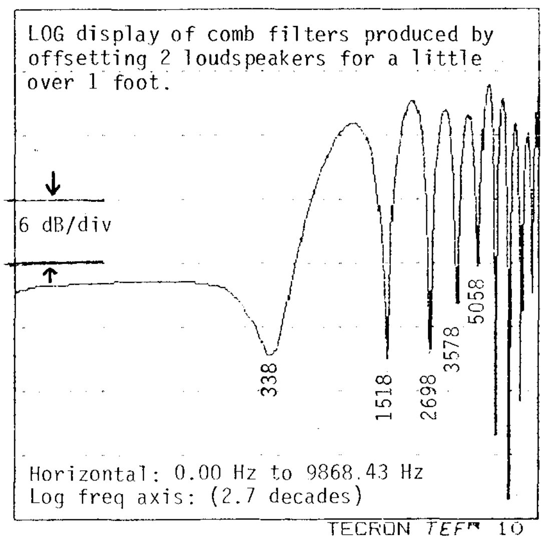


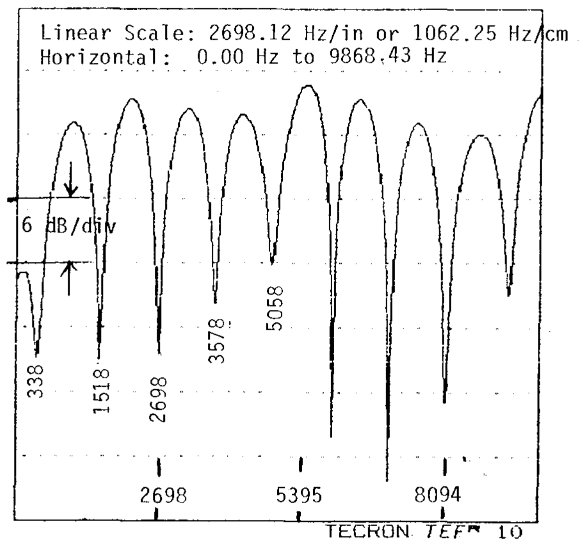


The second photo of RTA is "smeared" but it is reproduced here because it shows the grids. If we had a sound system with a raw house curve similar to this one, we would conclude that a little equalization would make it a perfect system. We know that the system has serious problems that are highly audible.







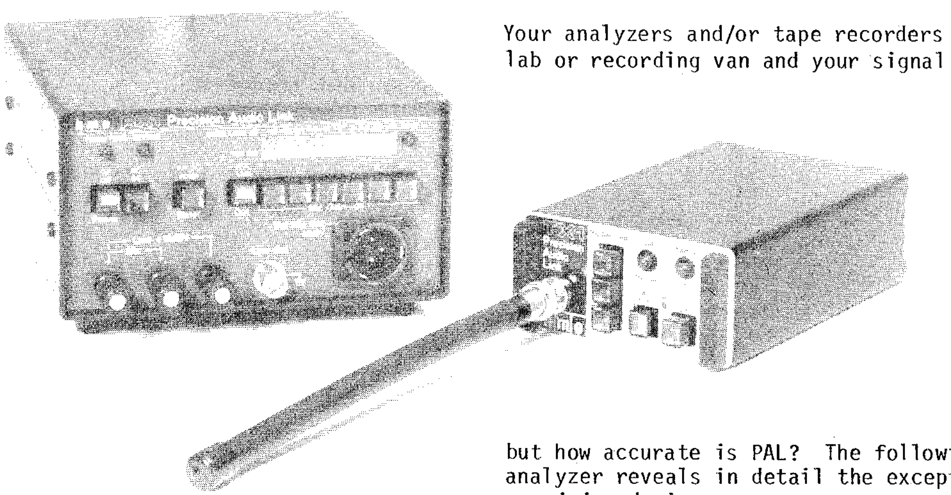


We have offset 2 loudspeakers by a little over a foot, which means that the comb filters are approximately 1180 Hz apart. Note that the bandpass filters on the RTA allow only a portion of the first comb filter to show - around 350 Hz. All the rest are obscured by the skirts of the analyzing filters on the RTA.

"PAL" - A PROGRESS REPORT

Those fortunate enough to have seen the "PAL" (wireless Precision Audio Link made by HME) in action have no doubt of its immediate, everyday usefulness. To be free of cables during a measurement session in all but the smallest of spaces provides an unheard of freedom in the choice of measuring locations and movement between them. How often have you, just as we have, wanted all your measurement equipment in one location, for example, near the control console in a large auditorium, only to find you couldn't do so because your cables wouldn't allow it?

Two PAL's allow the following advantage:



Your analyzers and/or tape recorders can stay in your mobile lab or recording van and your signal is sent to the console on one PAL channel and then picked up by a sound level meter and returned to the input of the analyzer by a second PAL on another channel.

We have recently converted our Bruel and Kjaer 1/4" microphone to B & K's very high quality preamp and battery-powered power supply so we can use it out in the auditorium with our PAL unit.

All this convenience is fine but how accurate is PAL? The following data taken on our TEF® analyzer reveals in detail the exceptional performance of this *precision* device.

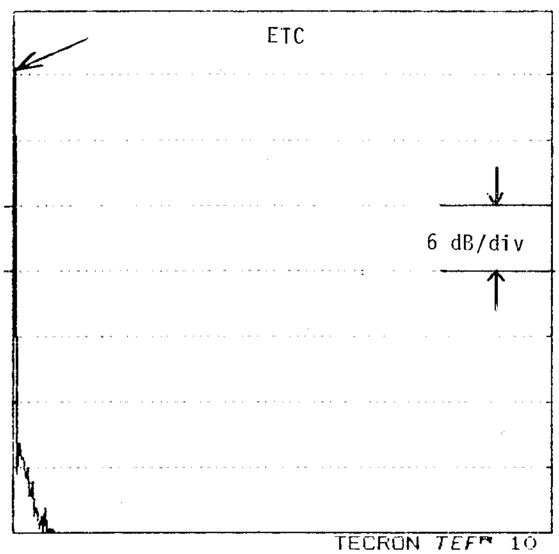


FIGURE NO. 1: The ETC plot reveals that its time response is far faster than any device we will have an occasion to measure *including very good electronic devices*.

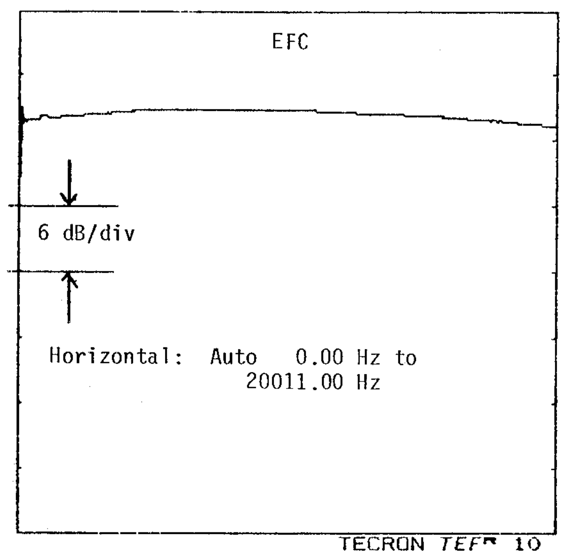


FIGURE NO. 2: The EFC plot is well inside HME's quoted specification and our extended experience with the unit has proven its stability as well as its uniformity of response.

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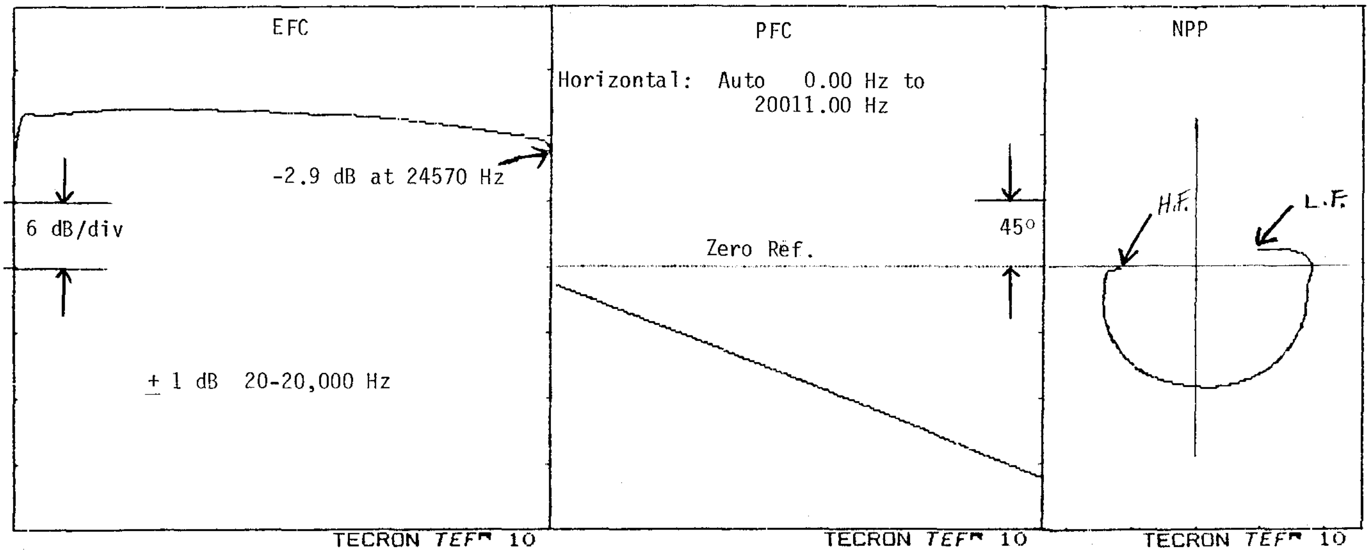
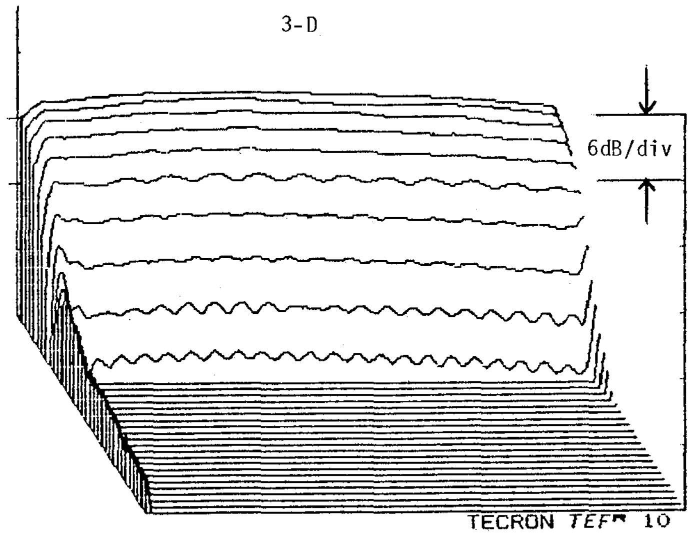


FIGURE NO. 3: A final EFC plot out to 25,000 Hz shows the extended range capability if needed.

FIGURE NO. 4: The PFC plot is what pleased us the most. We have an absolutely uniform phase response over the entire audio band with the lowest overall shift possible for the bandwidth concerned.

FIGURE NO. 5: The NPP again confirms the excellence of the engineering that has gone into this product.

FIGURE NO. 6: The "decay side" of the 3-D response has the same rate a piece of straight wire would exhibit for the analyzer settings used.



Conclusions

We have used our PAL to measure two of the three greatest concert halls in the world. We have used it in the United States and Europe. We have used it in R.F. saturated New York City and at the ranch for measurements a quarter of a mile away. We think that a serious audio measurement system is simply incomplete without it.

SYN-AUD-CON IN EUROPE - 1983

KLM direct from L.A. to Amsterdam. Immediate pleasant surprise - the car we had requested is unavailable, so we receive an upgraded model at the lower price. It turns out to be a Ford Sierra (made in Germany), due to be seen in the U.S.A. in 1985. It outperforms the BMW we had last year. Fun to drive in traffic during the pouring down rain at 185 KM/hr (approximately 115 miles/hr); stable as a rock; braking more than adequate; sensitive steering (hydroplaning easy to sense). We were passed only by Porsches, Mercedes 280s and up, Alfas, and other special spaghetti cars. Porsche Turbos pass you while you're doing 185 KM/hr, as if you've just had your engine fall out (some doing around 250 KM/hr.). Europe is still the automobile driver's paradise. Dutch Police don't stop you, they photograph you and you receive your ticket through the mail with your license plate clearly in the picture beside the radar reading at the moment of the photograph. The "Reich's Police" use Porsches for the rare chase that occurs.

While in Amsterdam, we witnessed the apprehension of a bank robber. We were next to him in our car when he was very suddenly surrounded by armed men, both in and out of uniform. The robber was on a powerful motorcycle. As we rounded a broad curve in a main Amsterdam thoroughfare, we saw ahead of us two small white police cars blocking off the street. We stopped beside the motorcycle. Just as we stopped, two Volvos rounded the same curve behind us, travelling in the center of the avenue, locked up their brakes and skidded about 100 feet. Plainclothesmen poured out of these two cars, brandishing semi-automatic pistols, and surrounded the motorcyclist. Fortunately, there were no shots fired. It reminded me of my childhood days in Chicago (circa 1932).

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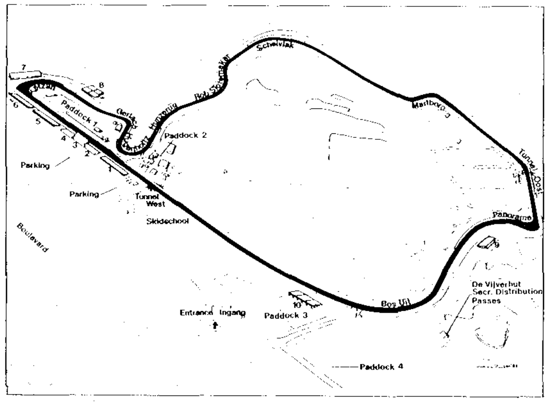
As if this weren't enough excitement, after we had completed the utterly fantastic Concert Hall Workshop, Carolyn, DON EGER and MARY GRUSZKA went with me to Circuit Zandvoort, the Dutch Grand Prix race track. On the strength of my misconduct on race tracks in the 1950's, we were allowed to use the track to turn some fast laps in the Sierra. Panorama was nearly my undoing, but hazy recollections of similar times in Porsches surfaced in time to shoot us out the exit of the turn, still on the track. Our grateful thanks to Tony Leggett of Ford who had the track for the day for vehicle testing and who allowed us the privilege of leaving our tire marks on its famous curves.

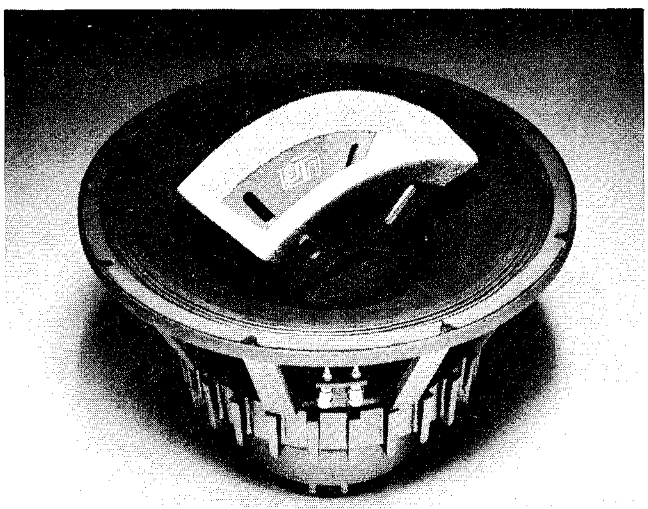


I haven't the time and space to tell all the adventures that occurred in our 1800 plus kilometers of travel, but I can testify it felt like I had stepped into a time machine and spun the dial back thirty years.

UREI ANNOUNCES THE 813B MONITOR LOUDSPEAKER

Syn-Aud-Con has made no secret of its approval of the UREI 813 monitor loudspeaker. Ever since its inception it has allowed practical comparison between time aligned™ and non-time aligned to the ultimate despair of the unaligned.

It is always with caution that we approach a "B" model of anything that worked well in the "A" version. Thus, we were particularly pleased to discover that the 813B, which we tested in the February TEF® Workshop, was an outstanding improvement in every way over the older unit. Believe it or not, it has a JBL tweeter that is not yet available in a JBL product. The woofer is from another supplier (not Altec). The 813B has the smoothest relative phase response with the least anomalies of any loudspeaker we have ever tested. Its Nyquist is an excellent approximation of the circle expected for the passband in question. It also *sounds* better to our ears than the previous model, exhibiting much easier-to-listen-to mid and high frequency response.



MODEL 801B COAXIAL LOUDSPEAKER

BRIAN OPPEGAARD (R) talking to JOHN WIGGINS of Community Light & Sound during our Loudspeaker Array Workshop. Brian, who works on the 813B project, was a student of Dr. Patronis' at Georgia Tech.

For the Tecron TEF® measurements on the Model 813B, see Figures 60 through 65 on pages 15 and 16 of the TEF® Workshop Tech Topic.

Syn-Aud-Con heartily recommends the UREI 813B for any serious control room monitor mission.

SMILE

Neckties strangle clear thinking.

LINEAR VS. LOGARITHMIC FREQUENCY SCALES

Both our G.R. 2512 FFT analyzer and our Crown TEF® System Ten allow us to look at the frequency spectrum with a *linear* or a *logarithmic* frequency scale. Since most of us have been carefully conditioned to use the log frequency scale and we feel familiar and comfortable with it, the use of the linear frequency scale can surprise us as it displays the frequency response of familiar devices.

Cut off filters, i.e., high pass and low pass filters, are designed to be (N) dB per octave, typical examples being 6, 12, and 18 dB per octave. 3-D TEF® measurements have shown us that "slope *rates*" are a key factor in the time behavior of the energy passing through a filter and that the significant "slope rate" is not (N) dB per octave, but (N) dB per Hz. Look at any TEF® 3-D plot of a loudspeaker viewed from the rear of the time scale, i.e., first sweep on the display was the last sweep in time and the last sweep (rear most) on the display was the earliest in time, and you will see "time smear" at the *lower* frequencies. Why? Because the slope rate of the system cutoff at low frequencies, which might indeed appear symmetrical with the high frequency cutoff slope rate at high freqs., is in actual fact much steeper *on a dB per Hz basis*.

FIGURE NO. 1:
LOG Frequency
Scale

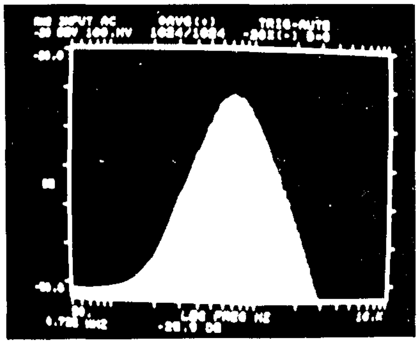
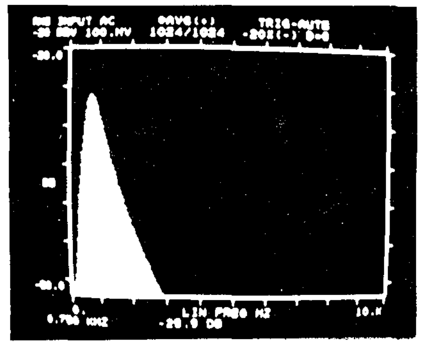
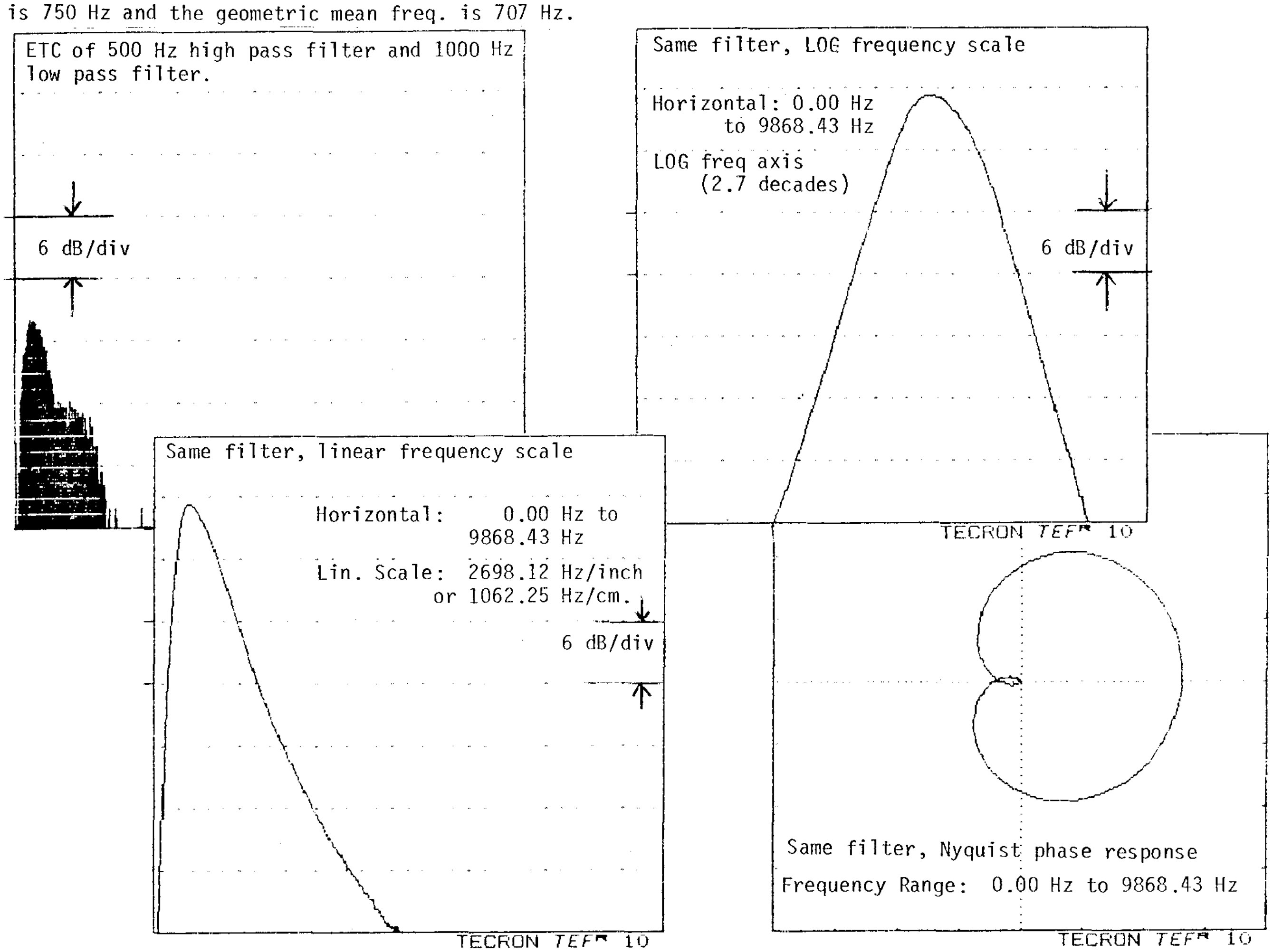


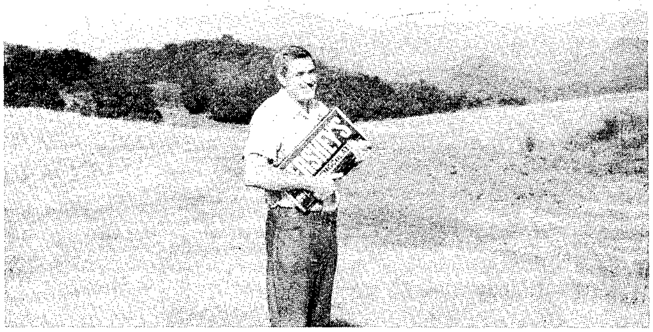
FIGURE NO. 2:
Linear
Frequency Scale



To illustrate the difference in these slope rates, we have photographed a low pass and high pass filter on both the lin. and the LOG freq. scale on our G.R. 2512. Both filters have the same dB per octave rate but obviously not the same dB per Hz rate. The H.P.F. is at 500 Hz and the L.P.F. is at 1000 Hz. The arithmetic mean freq. is 750 Hz and the geometric mean freq. is 707 Hz.



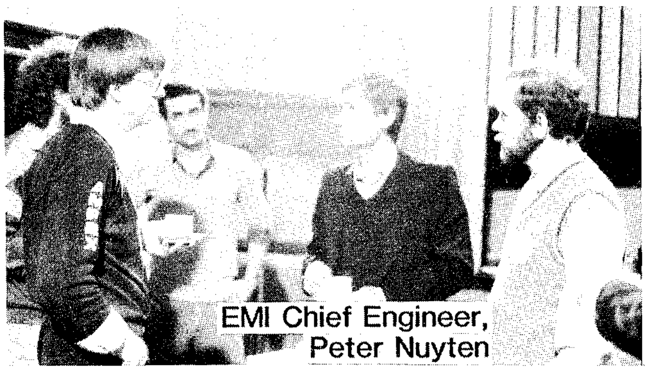
"SLIGHT OF HEART"



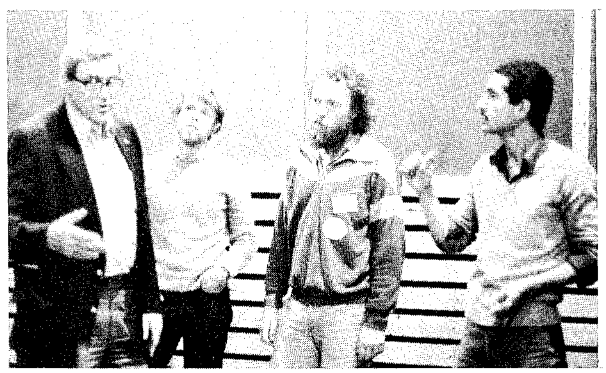
You may have noticed in recent pictures of Don in the Newsletter that he looks "*slightly*" (edited at his request) heavier. We believe that there is some "slight" connection between the picture shown here and the subsequent photos in the Newsletters. That's a 10 pound Hershey Bar he is clutching and the only other slight was his slight-of-hand act in making it disappear. It was a gift from Don's adviser on "shock absorbers."

STUDIO MEASUREMENT WORKSHOP AT EMI

We wanted to hold a one-day Studio Measurement Workshop in Amsterdam for our Syn-Aud-Con friends. Gerard Hali, Director of EMI Studios outside Amsterdam in Heemstede volunteered a control room and studio within his facility (cornerstone laid by Maria Callas).



EMI Chief Engineer, Peter Nuyten



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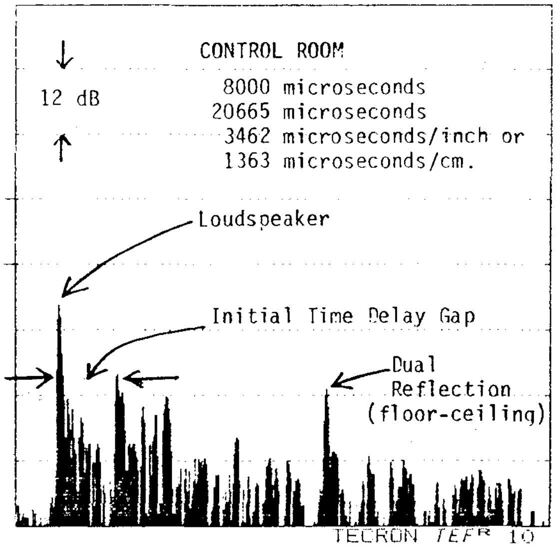
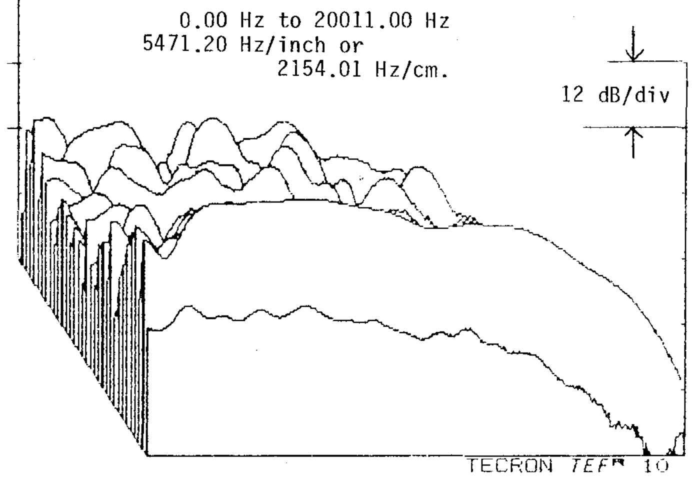
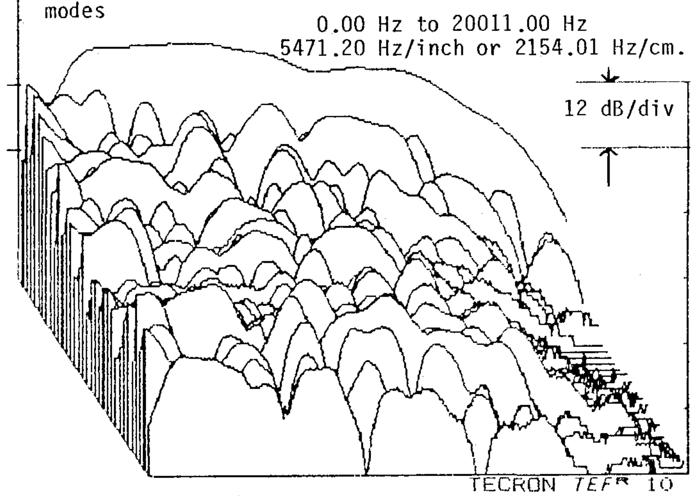


"Front" view of monitor response and early room modes



"Rear" view of monitor response and early room modes



We had one of our scariest experiences here. We opened the class by showing how the direction of reflection can be determined by passing a hand around the microphone and proceeded to show where each reflection was coming from *EXCEPT* one. It wouldn't disappear even when we used a large sheet of absorption.

The sweat began to pour. How do you tell the class, which had just started, that we had a reflection we couldn't locate? After about 15 minutes, cool heads prevailed and we moved the microphone and discovered the cause. Later we described the problem to Victor Peutz and he *instantly* diagnosed the problem -- a double reflection. The JBL biradial was mounted half-way between the hard ceiling and wooden floor near the loudspeaker. The console was open underneath and, at that position at the console (the mixer's position), there were two equal distance reflections joining into one energy reflection.

Remodeling was started on the EMI control room the day after our measurements. Construction was scheduled to begin before our workshop but Mr. Hali held off on the work until we made our measurements.

SYN-AUD-CON "CARE PACKAGES"

Going to the Post Office, a 30 mile round trip, or to our UPS pick-up, the local cat motel and a 40 mile trip, is always an adventure for Syn-Aud-Con employees. Wild animals abound down our 7 miles of private road, wild drivers abound on the 8 miles of the Ortega Highway, and a wild Southern California culture prevails all along our coastal plain. One has to personally visit a Southern California beach community to appreciate the variety of dress and undress encountered among the natives.

The greatest adventure of all, however, is the surprises awaiting us at the Post Office or UPS: Ten pound hershey bars, belt buckles, books, T-shirts, and other pleasant new experiences. In one week, we received two magnificent belt buckles from BOB PINKSTON, United Artists in Dallas, and ALLAN SEIPMAN of Taft Broadcasting in Houston. Bob sent the marvelous "State of Texas" belt buckle and Allan sent the beautiful "Screaming Eagle" Pratt & Whitney buckle. I wear the buckles in class on a belt made for me by one of our Disney grads, KEN GRUBER.

KATHY & CHARLES BILELLO, both Syn-Aud-Con seminar & workshop grads, from West Hempstead, New York, sent us a picture of their new baby, all decked out with a Syn-Aud-Con T-shirt. Charles designed and silk screened our logo on the T-shirt.

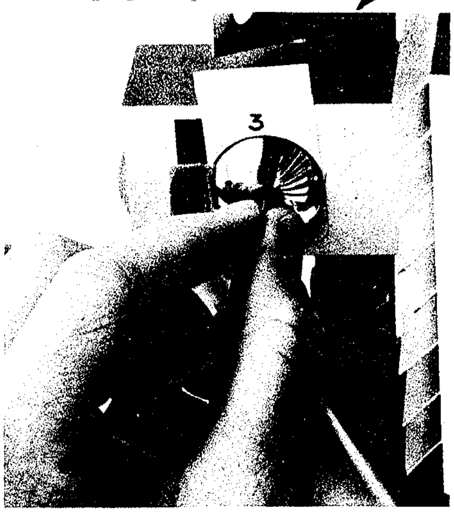
FORM FACTOMETER

In a letter from Doug Moss of Ram Audio in Alabama, he wrote:

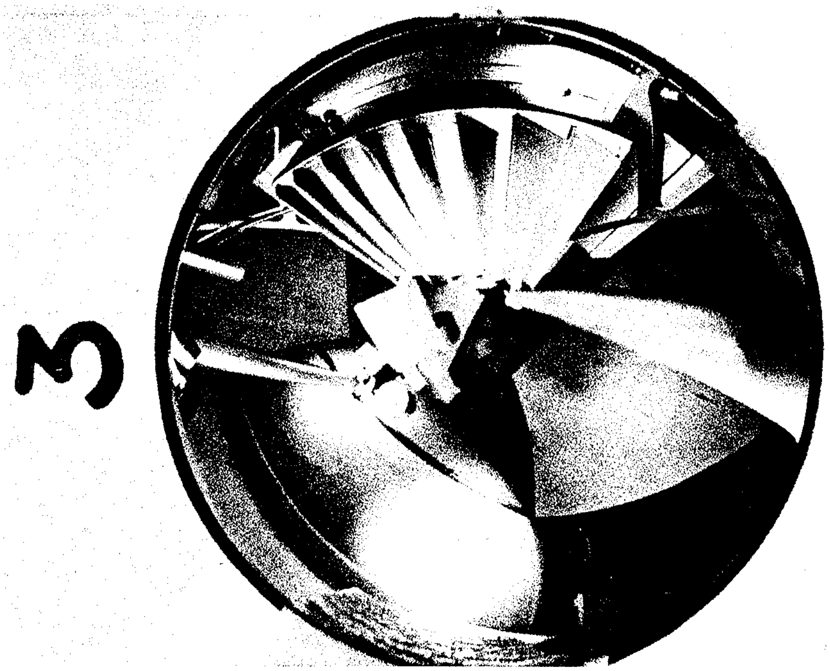
"The first name I gave you for an old mechanical view factor (form factor) device used in thermal analysis at NASA-MSFC and other places was incorrect - it is really a FORM FACTOMETER.

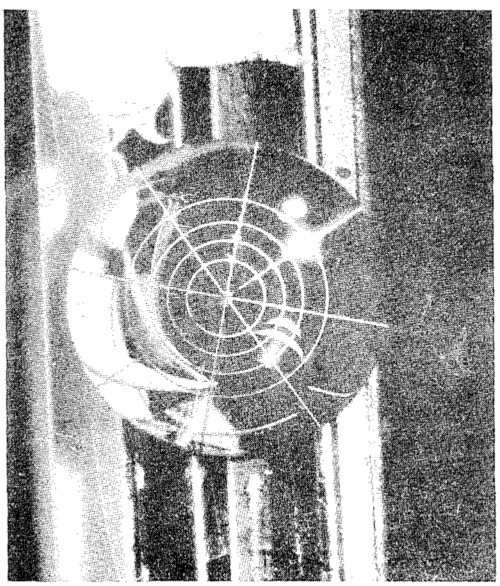
"The FORM FACTOMETER is calibrated in 'equal form factor' grid areas on the reflective surface of the spherical segment when viewed straight down onto the device (by eye or camera). This, as you said, would be analogous to "Q" (actually constant/Q). One aid, I believe, in the situation where architectural scale models are used, would be to color code areas known to be in an acceptable band of distances wherein the inverse square law doesn't give you too much of a problem. Several colors representing each preselected area would be present in the architectural model. These colors would show up on this device to aid in array design."

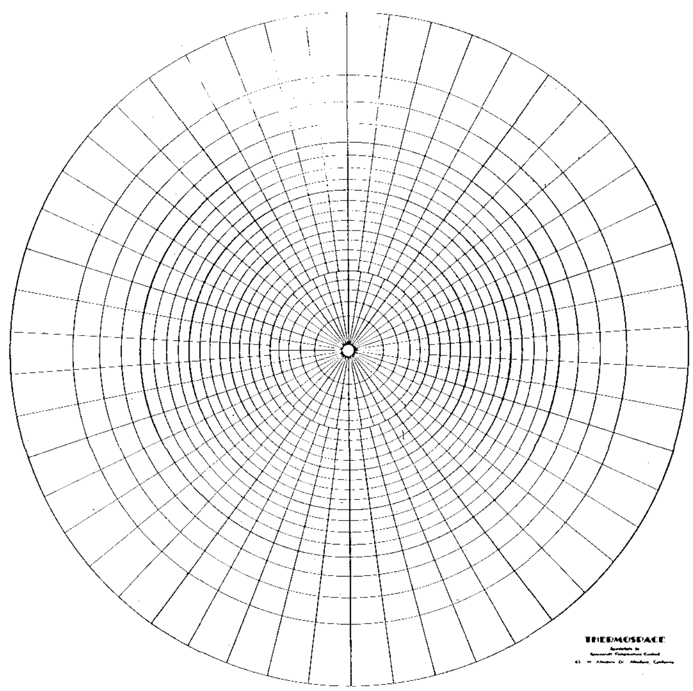
Attach Form-Factometer to the surface. Always handle the instrument with the handling spike provided.



Enlarge the photograph of the Form-Factometer to the diameter of the overlay grid.







That percent of the *energy radiated* from a diffuse surface which is intercepted by another surface or group of surfaces *is defined as a radiation form factor*. This factor relates the position and size of the surfaces exchanging radiant energy.

Continued next page.....

The Form Factometer is a precise optical instrument which has been designed and constructed specifically for the purpose of evaluating this factor in a practical manner. When properly mounted in a scale model of appropriate size, the "Form Factor" is equal to the number of "grid areas" (see illustration of grid) counted, multiplied by 0.025.

It seems to us that this clever and useful device could be directly applied to the study of the diffusion of rear walls in scale models of LEDE™ control rooms and to the measurement optically of the preferred Q at difficult architectural intersections on scale models of concert halls, etc.

We are grateful to Doug Moss for having brought this to our attention.

AES CONVENTION - EINDHOVEN 1983

At any given AES Convention, the number of significant papers rarely exceeds the fingers on one hand but it's well worth the time spent in listening to the obvious being rehashed in order to identify those with a new idea. Even more important than the papers is the data that can be picked up in conversation with talented individuals, some of whom will never give a paper, who can impart invaluable insights into product development and usage.

The Dutch are a delightful people with a sense of freedom rarely found. Their genial disrespect for authority seems not to have a dark side but stems from an ingrained "live and let live" toleration of just about anything short of anarchy. It is not difficult to understand how the Dutch became the haven for the persecuted during Europe's religious wars.

The AES Convention was held in the POC Convention Center owned by Philips. The POC has restaurants, a concert hall (complete with electroacoustic enhancement of the acoustics - definitely in the development stage) and a huge convention hall complete with side area demonstration rooms.

Outstanding AES Experiences

The introduction of the Tecron TEF® analyzer to Europeans had even more than the expected impact. Many in Europe have been devoting serious theoretical effort towards the problems the TEF® so easily measures and thus they were equipped to appreciate the breakthrough this remarkable analyzer represents.

Philips demonstrated an outstanding use of the basic Bessel function equations applied to the design of a multispeaker array, wherein the functions are achieved through choice of impedances, series or parallel connections, and in and out of polarity connections. A twenty-five speaker array per channel was demonstrated amplifying a live group and then in playback of the same group. This very practical and excitingly effective technique can be applied to microphone arrays as well.

Two extremely gifted investigators from the Philips Research Laboratories, Cornelis P. Jause and Arie J. M. Kaizer, have extended the published data on their use of the Wigner distribution. This new data was published in the new AES Journal.

A company called Holophonics had a "black box" microphone demonstration (a circle transducer worn around the head like a headband) that provides a listener wearing headphones the illusion that he can hear up and down, front and rear, as well as side to side.

Bruel and Kjaer displayed their new "studio" microphones, types 4003, 4004, 4006 and 4007. Of great interest to us is their photographs of its use turned upside down and spaced very close to a large surface. They also demonstrated a very versatile 2 channel FFT, the type 2032. So far as we are aware, this is the first FFT to include the Hilbert as well as the Fourier transforms. Price is around $30,000.00.

We had the privilege of meeting Dr.-Ing. Wolfgang Ahnert of East Berlin and to receive from his hands a copy of his "Akustik in Kulturbauten" (Acoustics in Culture Halls). Dr. Ahnert is also the co-author, along with Prof. Dr.-Ing. Walter Reichardt of "Grundlagen der Beschallungs - Technik" (Fundamentals of Sound Systems). Dr. Ahnert has been involved with systems in 6000 plus seat auditoriums using up to 800 loudspeakers to generate and control the level and timing of direct, early reflected, and statistical reverberant sound fields. We were pleased to find he spoke excellent English (as well as six other languages) and we are curious to hear from those of you that would have an interest in meeting with such a scientist in a special seminar if he were to make a trip to the U.S.

We were very pleased to witness the prosperity of Europe and were reminded once again how so many diverse cultures, living in such close proximity to each other without common national purpose, generate remarkably innovative approaches to what in the United States has become standard practice.

SMILE

Churchill was told by one of his Generals, whom he has been prodding to study logistics, that "familiarity breeds contempt." Churchill replied, "Without some familiarity, we wouldn't breed anything."

USING THE TECRON SYSTEM 10 TEF® ANALYZER TO MAKE 3-D POLAR PLOTS

Normal polar plots are made either one frequency at a time or one band of frequencies at a time (i.e., 1/3 or 1/1 octave bands). They are plotted either on circular "polar" paper or as a series of frequency response curves on a level recorder chart.

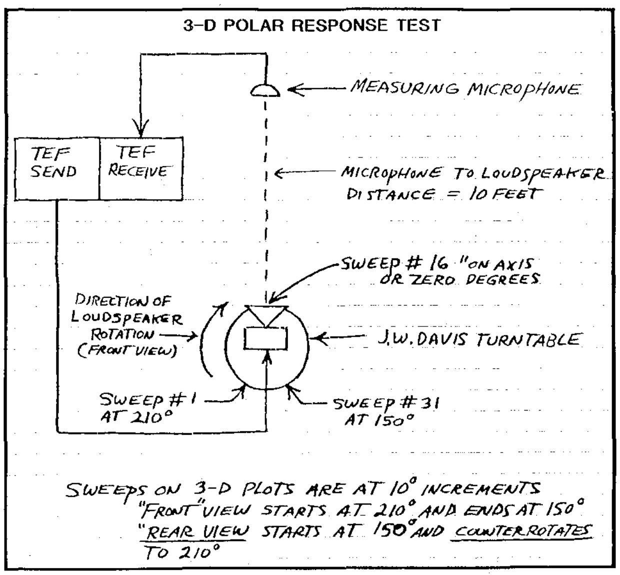


FIGURE NO. 1

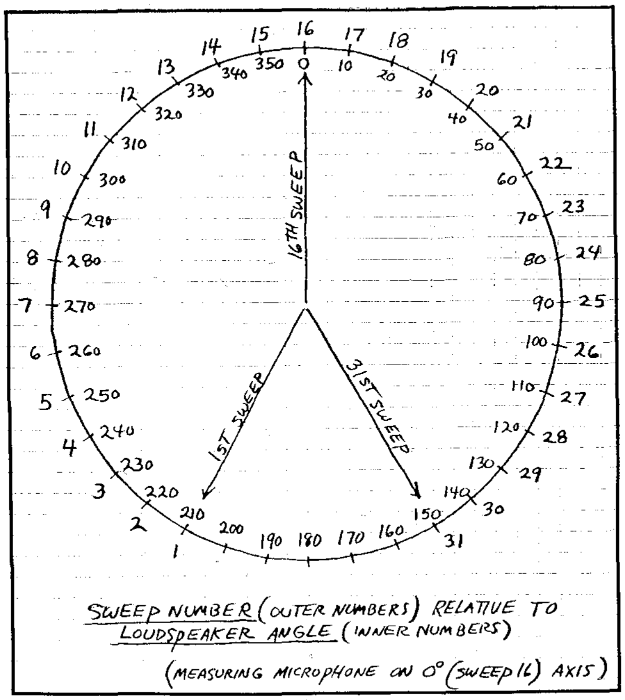


FIGURE NO. 2

The Tecron System 10 TEF® (Crown/Tecron trademark) analyzer allows a three dimensional view of angular coverage patterns by means of its 3-D mode of analysis. The 3-D mode makes 31 consecutive sweeps. If sweep number one is made at 210° (0° is on axis), then there are 15 steps of 10°/step to reach 0° in time for the 16th sweep followed by 15 more steps of 10°/step until you reach the 31st sweep at 150° (see Figure No. 2).

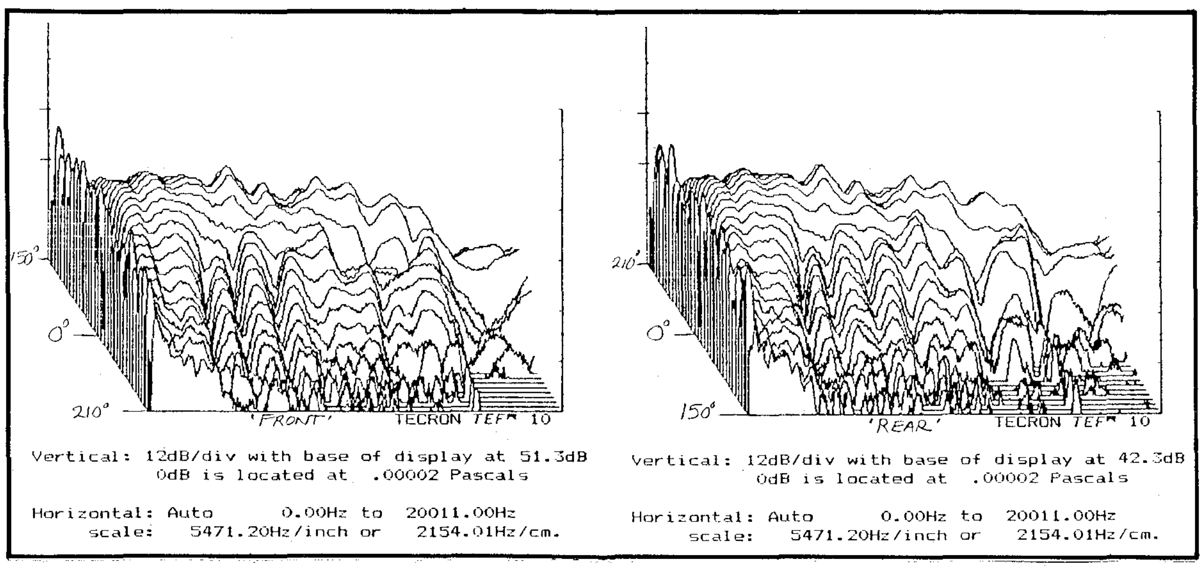


FIGURE NO. 3

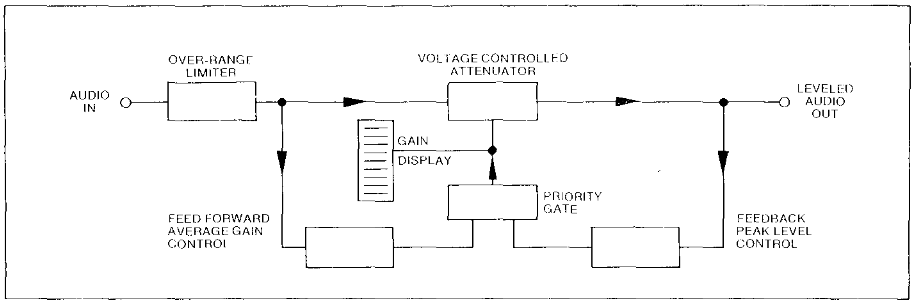
The 3-D mode normally is set up to "step" the time window by some chosen time increment between each of the 31 sweeps. In making 3-D directivity plots (3-DDP), the Tecron TEF analyzer is instructed to make all 31 sweeps at the same "time offset" (i.e., the offset associated with the direct sound measurement). Thus, the only difference between the 31 sweeps is the angle between the loudspeaker's axis and the measuring microphone.

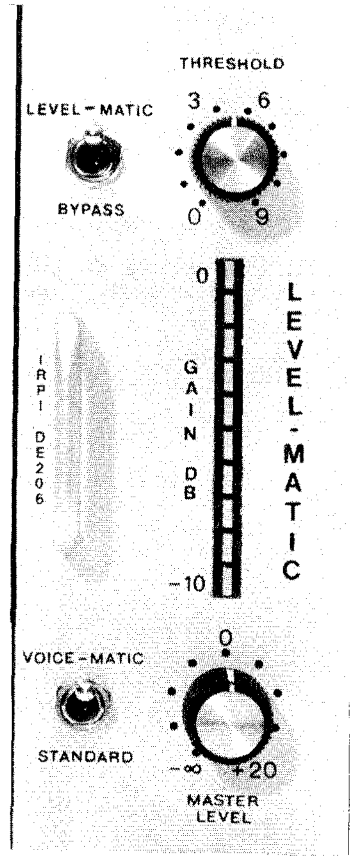
INDUSTRIAL RESEARCH PRODUCTS, INC.

Industrial Research Products, Inc. (IRPI) is a long time Syn-Aud-Con sponsor and our first choice, without question, when we require a conventional digital time delay. (We define *"conventional"* digital time delay as those units with steps in some multiple of milliseconds as opposed to special purpose digital time delays for array alignment, etc.)

The IRPI mixers have thoroughly proven themselves as the mixer leading the way and they have now added a new "Level Matic" Master Module (the DE-206) as a further enhancement of their product.

LEVEL-MATIC™ FUNCTIONAL DIAGRAM

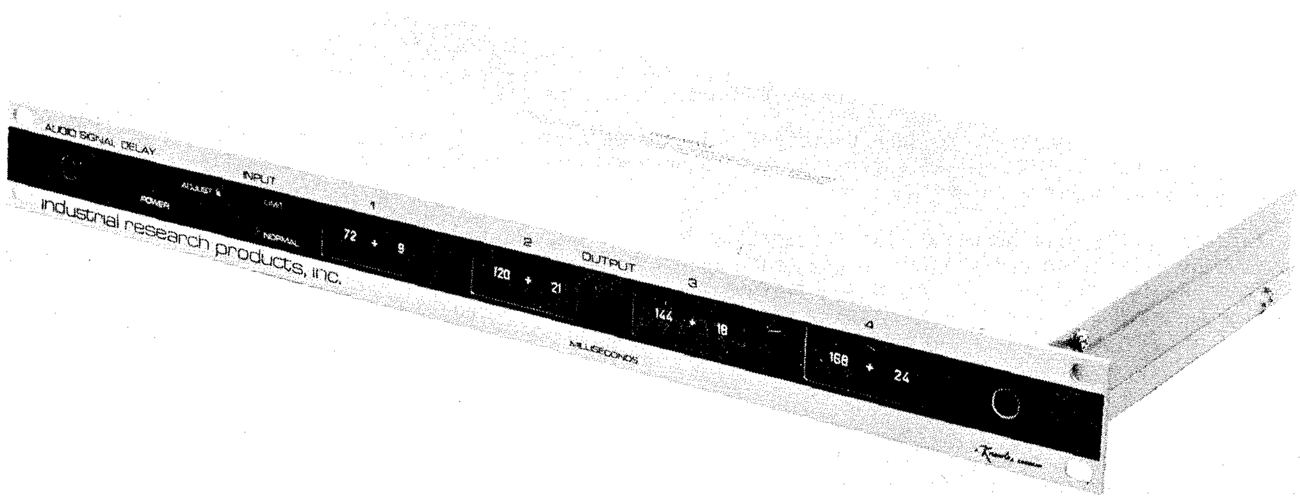




SPECIFICATIONS*

Level-Matic DE-206:

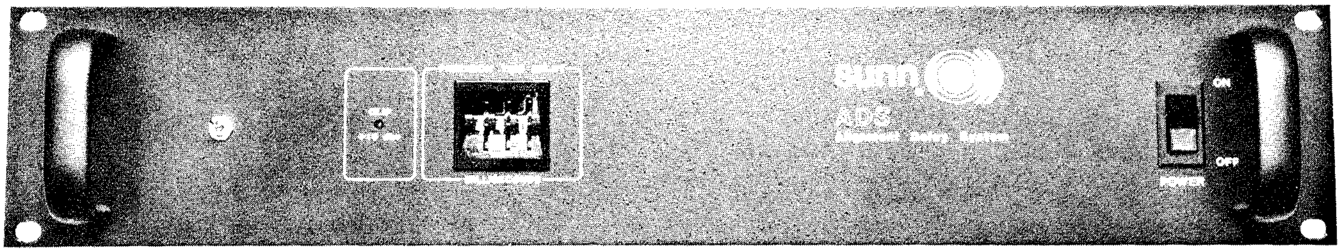
Gain Range	10dB
Gain Display	10 step LED bar graph
Frequency Response	± .5dB, 20Hz – 20kHz
Distortion (THD + noise)	less than .15%
Noise (Typical control settings	–70dBA @ threshold
on DE-4013 and DE-206)	–80dBA @ 10dB gain reduction
Feedback attack/release	100msec/500msec
Feedforward gain slew rate	1dB/sec nominal (.3dB/sec to 3dB/sec internal adjustment range)
Over-range limiter threshold	25 dB above Level-Matic™ threshold
Over-range limiter distortion	less than 1% @ 10dB limiting
Maximum limiting	15dB



Syn-Aud-Con graduates are heavy users of IRPI products and report back to us their satisfaction with them.

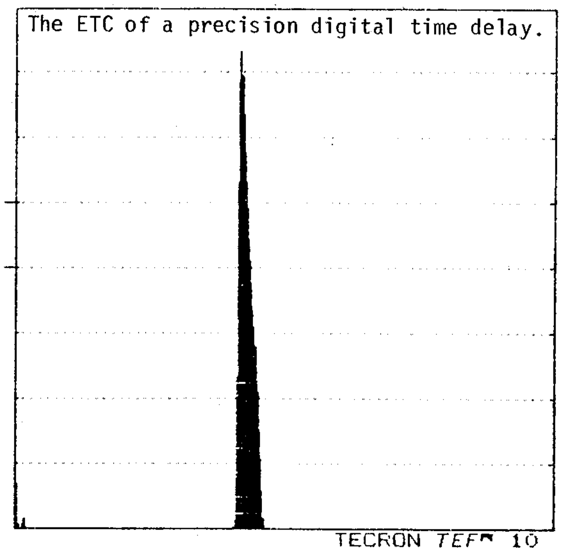
A REMARKABLE DIGITAL TIME DELAY FROM SUNN

Recent Syn-Aud-Con graduates are aware of our desire for a precision digital time delay that would allow us to "align" multiple horn systems in the region where their coverage patterns meet and partially overlap. The Sunn ADS Time Delay System meets this need with performance to spare.



A Unique Accomplishment

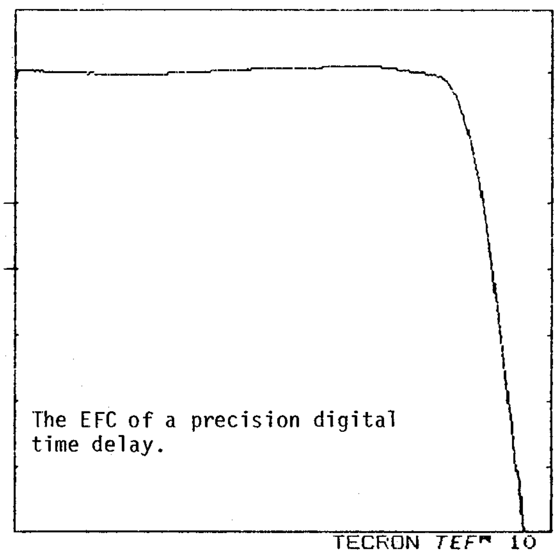
To date, this unit is the sole digital device we have tested to be completely free of high frequency "ringing" on our 3-D measurements. That, in our experience, is a totally unique accomplishment worthy of very special notice.



The ETC of a precision digital time delay.

Vertical:
6dB/div with base of display at 96.3dB
0dB is located at .00002 Pascals

Horizontal: 0 microseconds or 0 FT. to
12665 microseconds or 14.3115 FT.
scale: 3.9129E+00 FT./inch or 1.5405E+00 FT./cm.
3462 microseconds/inch or 1363 microseconds/cm.



The EFC of a precision digital time delay.

Vertical:
6dB/div with base of display at 84.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 24945.20Hz
scale: 6820.26Hz/inch or 2685.14Hz/cm.

We have both used and measured the Sunn prototype unit and were excited to discover how audible a mere 20 usecs (that's right, *MICRO SECONDS*) misalignment is. The Sunn unit is a delight to use and a real eye and ear opener in terms of adjusting large arrays.

We are told the price will be reasonable (especially for what amounts to a Rolls Royce unit). In addition to its obvious benefit to an array builder, psychoacousticians can now have a precision tool with which to examine small misalignments in time. The marvelous freedom it provides an array designer in allowing a wide choice of mounting geometrics that can be acoustically corrected via this delay should quickly lead to being able to put two signals in the same place at the same time without having to do the same with two unwieldy horns.

Sunn ADS Time Delay System

The Sunn ADS is a high precision digital time delay designed for achieving time-alignment in professional sound systems. The ADS differs from other commercially available time delay systems primarily in its ability to generate the very short time delays (typically 10 to 1000 microseconds) needed to align speaker systems. It is also free from the frequency response limitations, excess noise and high distortion common to other audio time delays.

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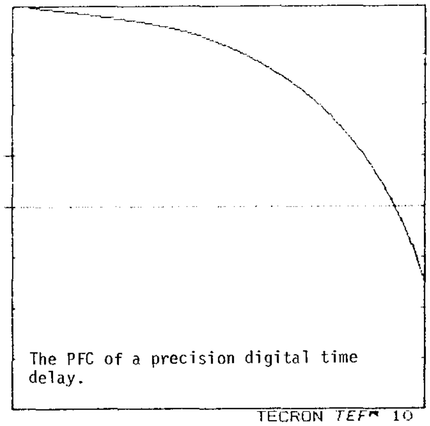
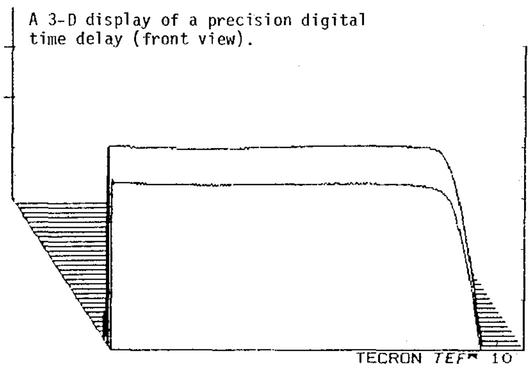


The PFC of a precision digital time delay.

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

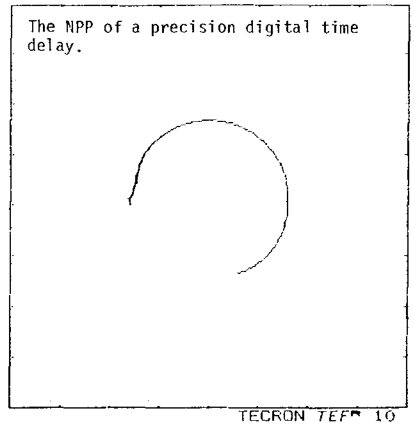
Horizontal: Auto 0.00Hz to 20011.00Hz
scale: 5471.20Hz/inch or 2154.01Hz/cm.



A 3-D display of a precision digital time delay (front view).

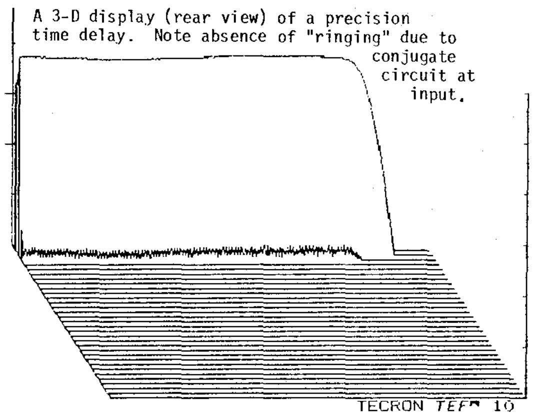
Vertical: 6dB/div with base of display at 111.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 24945.20Hz
scale: 6820.26Hz/inch or 2685.14Hz/cm.



The NPP of a precision digital time delay.

Resolution: 9.0228E+00 FT. & 1.2524E+02Hz

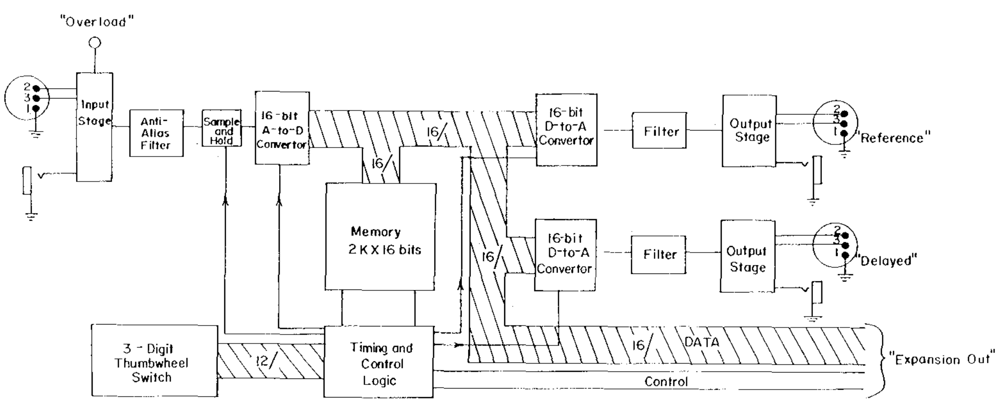


A 3-D display (rear view) of a precision time delay. Note absence of "ringing" due to conjugate circuit at input.

Vertical: 6dB/div with base of display at 111.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 24945.20Hz
scale: 6820.26Hz/inch or 2685.14Hz/cm.

The ADS has two outputs: a "Reference" output and a "Delayed" output. The signal at the "Reference" output will be delayed slightly with respect to the input signal due to the intrinsic (and unavoidable) propagation delays and conversion times of the unit. The signal at the "Delayed" output will see these same time delays *plus* the desired time delay chosen by the thumbwheel switch on the front panel. Thus, the time delay *difference* between the two outputs will be exactly equal to the front panel setting. In a typical installation the speakers for which delay is desired would be connected to power amps driven from the "Delayed" output.



BLOCK DIAGRAM – SUNN ADS

Continued next page.....

The normal, or undelayed, speakers would be driven by power amps connected to the "Reference" output. This configuration allows the relative time delay between the two speaker systems to be adjusted to any value between 0 and 9.99 milliseconds, in 10 microsecond steps. In spatial terms, 10 microseconds is equivalent to about one-eighth of an inch. Thus, the maximum delay of 9.99 milliseconds corresponds to a physical distance of approximately ten feet.

The ADS also has a digital output port which will allow the connection of expansion modules. These modules will allow low-cost extension of the system to multiple independent delays and will make possible longer individual delays.

Features

Time delay to 9.99 msec in 10 usec steps

20 kHz bandwidth

THD less than .05%

Crystal-controlled clock for extremely low drift

Digital expansion output

Balanced and unbalanced inputs and outputs

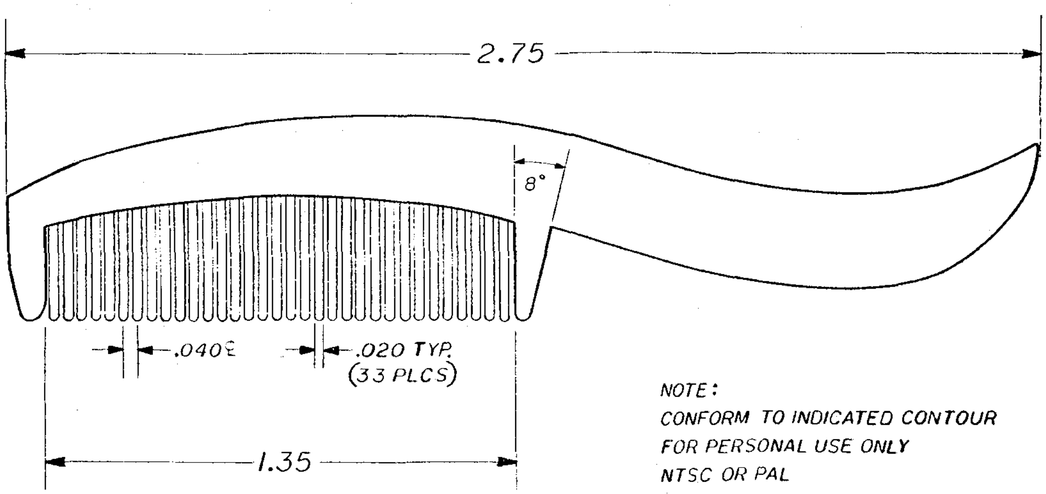
Input overload indicator

Optional anti-tampering cover

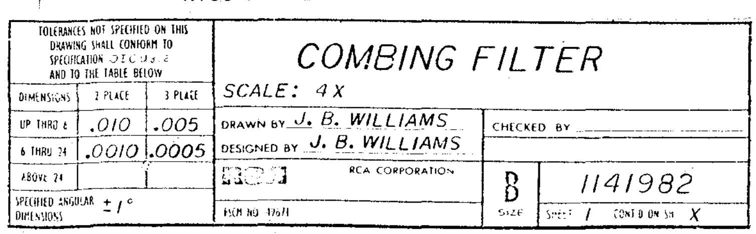
Specifications

Relative time delay:	0 to 9.99 msec in 10 usec steps
Method of delay adjustment:	Thumbwheel switch - 3 digits
Delay accuracy:	Better than 1%
Delay stability (drift):	Less than .01%
Intrinsic delay to reference Out:	120 usec nominal
Frequency Response:	+0 -3dB, 5Hz to 20 kHz
THD:	Less than .05%, 20Hz to 20 kHz
IMD:	Less than .05%
Signal-to-Noise, A wtd. from full out:	Better than 90dB
A to D resolution:	16 bits
Sampling rate:	50kHz
Input Impedance:	10K ohms
Output Impedance:	150 ohms
Maximum Input Level:	10V RMS
Maximum Output Level:	10V RMS into 600 ohm (t22 dBM)

COMB FILTERS



Courtesy of BOB HAGENBACH
RCA, Indianapolis, Indiana




Anaheim - 1983
San Juan Capistrano - 1983

REMOTE POWER CONTROL SYSTEM

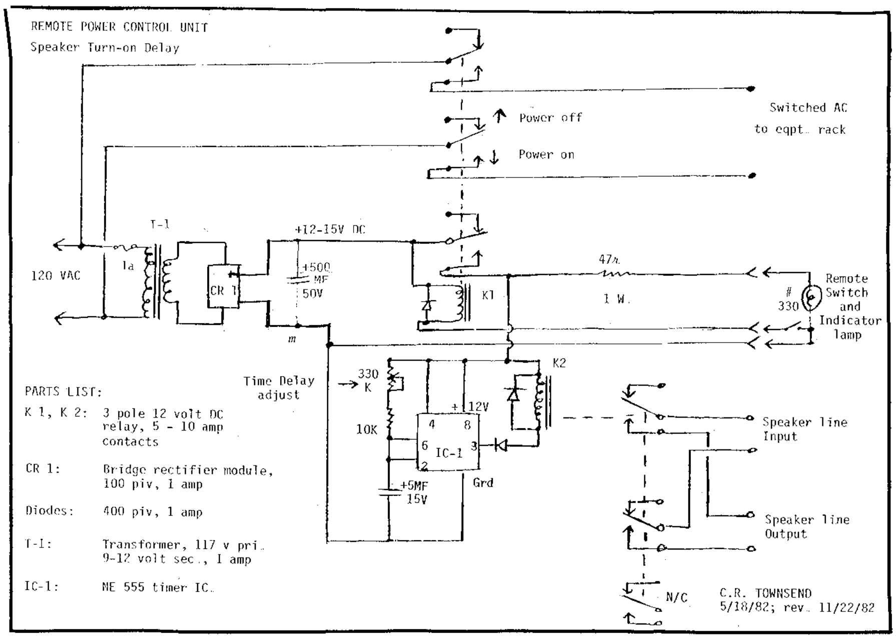
CHARLES TOWNSEND of T-Comm Systems in Tallahassee, FL., has sent us another of his clever circuits.

I am in the process of upgrading the sound system down at the Church (St. Paul's United Methodist Church) and thought you might be interested in how I solved a problem at St. Paul's.

The system operator's position is in the balcony at the rear of the church and will remain here following system renovation. The equipment is located in a small room behind the main sanctuary, which makes it a rather inconvenient place to run to to turn the system off and on for each of the 3 services held on Sundays. What was required was:

1. *A remote power control system which could be operated from the balcony.*
2. *A time delay circuit which would allow the system to power up, stabilize, and then apply energy to the loudspeaker system. The existing tube-type amplifier takes care of this now, but we were concerned about the potential turn-on 'thump' which could occur with the new 'instant-on' solid state equipment when we upgraded.*

The enclosed circuit shows how we solved both problems.



When the remote switch is closed, relay K-1 pulls in and routes 120 VAC through to a pair of power outlets which feed the equipment in the rack. The lower set of K-1's contacts puts +12 volts on the 555 timer circuit (IC-1) which begins a 3 second timing cycle, charging up the capacitor on pin 2. When the required threshold is reached at pin 6, the output (pin 3) goes to a 'low' state, which then energizes K-2. The speaker lines, which are routed through the contacts of K-2, are then connected to the amplifier output. When the remote switch is opened, K-1 drops out, then K-2, and the circuit is ready for the next power-up cycle. The power supply (T-1, CR-1 and the 500 mf filter capacitor) remains powered at all times to provide the +12 volts required by K-1.

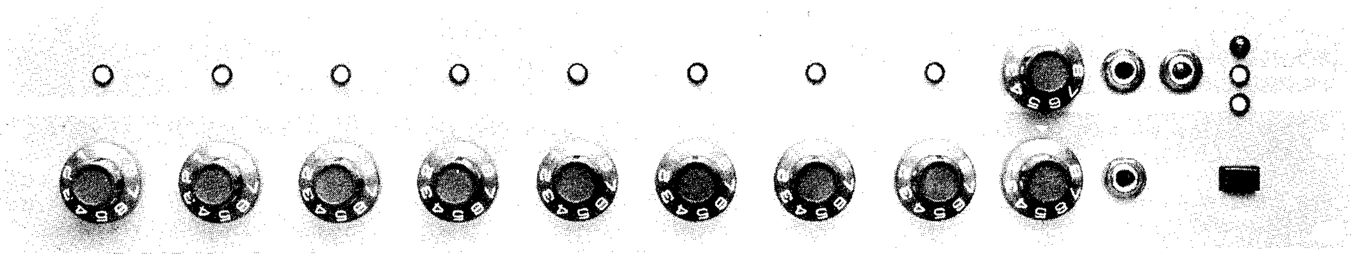
The entire circuit was mounted on the wall near the new rack location in a metal JIC type cabinet. The wiring for the remote switch and indicator lamp was obtained by using three wires in an existing unused cable which was left over from the original 1957 installation.

The relays are standard industrial types (bought surplus) and are 4PDT with 12 volt, 60 ma coils. Other types, such as Square-D KPD-12, should also work. The user should verify, in any case, that the contact ratings of K-1 are heavy enough to carry the load imposed by the sound equipment.

In case you are wondering about the mike mixer (located in the balcony), it is powered locally by an AC outlet and is turned on first by its own switch.

SHURE AMS8000

One would have to be very remote from audio today not to have read about the new Shure automatic mixer and "smart mic" system. Shure's beautiful color ads are on the back cover of practically all the audio magazines.



Shure made a dramatic entry into sound reinforcement with an "integrated system" - the Shure Vocal master in the 1960's. The AMS is an integrated system of a mixer and microphone that work together and only together.

The integrated system is both its strength and its weakness. The "strength" is that each AMS microphone is activated only by sound that originates within a 120 degree window of acceptance. Sound sources outside this window will not make the microphone turn on, regardless of their loudness. The "weakness" is if you cannot or will not use the Shure smart mic - if your client won't allow microphones on the conference table. That weakness is the only negative we have heard from the field. All other comments have been positive.

Shure tells us that the AMS is the first of a series of new products - products they feel will have the impact of the AMS.

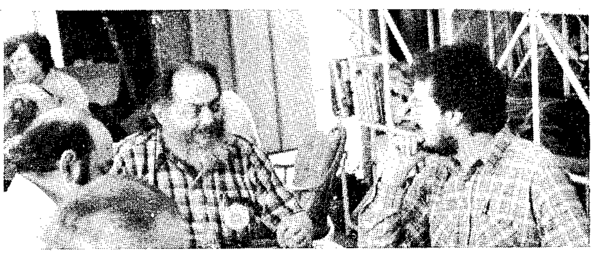
J. W. DAVIS & COMPANY GIVES 5% CREDIT

J. W. Davis & Company has announced that it will apply a 5% credit toward attendance in a Syn-Aud-Con seminar or workshop.

This means that *any* product ordered from J. W. Davis & Company will accumulate this credit. For instance, a $1,000.00 order from J. W. Davis will mean $50.00 applied toward the $395.00 registration fee, and $7,900.00 would mean that J. W. Davis will pay your full registration fee.

It will be necessary to notify us or the Syn-Aud-Con representative in the area where you wish to attend a Syn-Aud-Con class, as attendance will be limited.

TEXAS TALES



ROLLIE BROOK (left) talking to BRIAN OPPEGAARD of UREI.

ROLLIE BROOK of Bolt, Beranek and Newman in Los Angeles is a fascinating and experienced acoustical consultant. When he found out that I collect memorabilia from Texas, he quickly added four books on Texas to my library on the subject. One of these volumes is labeled *"Texas - A World in Itself"* by George Sessions Perry and contains a number of handwritten pencil notes in the margins where some Texas reader disagreed with the author or desired to add editorial comment.

At one point, Lyndon Johnson is mentioned and penciled in beside his name is the following note:

"They through away the child and kept the afterbirth, naming it Lyndon." (Spelling as read.)

Laugh as you read it but remember Texans have a fondness for their villians as well as their heroes and I suspect it's catching, witness J. R. Perry who points out that *"A Texan who wishes to perpetrate business frauds and pays off discreetly may succeed.....But if an indiscreet soul branches out into the realm of physical violence, like Chicago racketeers, it is not well for him to take up permanent abode in Texas. Police may be purchasable, half the sheriffs tempted, but when it comes to the Texas Rangers, their competitive spirit will have been aroused. His breach of the civil peace they will regard as an attack upon their dignity as guardians of that peace. No longer are bribes acceptable. Their pride, their belief in the legend that they are the guttiest men on earth, has become involved. And, with unerring certainty, they'll cut him down."*

The real Texas is somewhere about 900 miles west of Texarkana and 1,000 miles east of El paso and certainly in the consciousness of all who enjoy legends.

MORE ON "SOME EXPERIMENTS WITH TIME"

DAVID CLARK's Tech Topic, "Some Experiments With Time," has generated truly fascinating mail here at Syn-Aud-Con. We reproduce three letters here. David has triggered real Synergy. The correspondents confirm again that there really are a lot of very active thinkers out there in the audio world.

Jack Wrightson, of Joiner-Pelton-Rose, Dallas, Texas, writes:

"I was pleased to read the paper by David Clark (Tech Topics Vol. 10, No. 5, 1983) concerning the audibility of time offset between drivers in a loudspeaker system. This letter is not to take issue with Mr. Clark's arguments (I agree with virtually all of his points) but rather to add some comments of my own.

"Any test of audible differences must take into account two factors: 1) the listener's degree of experience with the task, and 2) the increased ease of detecting differences in changing (or adjustable) signals when compared to static signals (i.e., swept sine waves vs. sequentially presented tones).

"Factor No. 2 typically lowers the threshold of audibility for comparisons in any experiment utilizing variable stimuli.

"We know well what the effect of time delay is on the spectrum or, say, a summed noise signal. However, in Mr. Clark's experiments, the two signals feature different pass bands and radiation sources. This will no doubt affect the depth of the comb filter induced notches in frequency response. Also, due to the lack of a true summation of the two signals, we actually have little knowledge of what the signal reaching the listener's ear was. It might have been better to filter and delay the signal and sum it prior to reproduction through a single loudspeaker. I have no doubt that frequency response errors are the major source of detectable differences. It is well known that, perceptually, the spectral content of a sound is its most salient feature (0.1 dB spectral differences are easily detectable by trained listeners). Psychoacoustic researchers are constantly searching for stimuli that can separate spectral cues out from temporal and phase cues, as spectral cues tend to dominate a listener's perception.[1]

"Mr. Clark's nominal 40 microsecond threshold should not be surprising as frequency aberrations should begin lower than the first notch occurring at 12.5 KHz with a 40 microsecond delay.

"Mr. Clark's comparison of the all pass filter stimuli to the straight delay stimuli is a classic case of apples and oranges. In the former case, the listener is presented with a complex series of frequency dependent delays. The complexity of this signal obviously must be confusing to the time discriminating abilities of the ear/brain system. It also points out a possible spectral explanation for your oft cited dictum on time delay between drivers (paraphrased) 'way out or dead on time alignment'. A very long delay, say 50 msec., will produce comb filtering at 20 Hz intervals. This type of spectral aberration is roughly uniform across the ear's critical bandwidths (approximately 1/3 octave spacing) and is much less noticeable (objectionable) than a single notch in the audio band whose effect spans over more than one critical bandwidth.[2] This helps explain why a close, but not correct, time alignment may sound 'worse' than one that is well out of alignment. It also explains why smooth, extended frequency response aberrations in sound systems are more noticeable than a very narrow, deep notch.

"Mr. Clark states that two loudspeaker systems with identical amplitude response, yet different time alignment, will sound essentially the same. This is undoubtedly true for a single system, however, in a stereo pair even a very small time delay (i.e., 7 microseconds) will produce spatial localization cues that may be audible to a trained listener. For an audiophile, this would result in a shift and/or wander of the image between his two loudspeakers. The nonaligned speaker system should also produce changes in sound quality along the same axis that its drivers are oriented.

"In summary, I wish to congratulate Mr. Clark on his efforts. I also hope to point out that experimentation must be very carefully conducted, especially in regard to stimulus generation and in terms of dismissing the audibility of any potential cues affecting sound quality.

"[1]Warren, R.M., Bashford, J.A. Jr., Wrightson, J.M.; Infrapitch Echo, Journal of Acoustical Society of America, 1980, 68(5) 1301-1305.

"[2]Fryer, P.A.; Intermodulation distortion listening tests, Proceedings of A.E.S. 50th Convention, London, March, 1975."

We also received comments from JIM FULLMER, of Acoustical Engineers Inc., Salt Lake City, Utah:

"I read with great interest the article by David Clark, 'Some experiments with Time' (Tech Topic 10:5).

"His ability to get to the bottom of a problem, at least as it relates to the final judgment of the ear, is admirable. Mr. Clark's article in the AES Journal (May '82) should be required reading for the Highly Inflated Phidelity Esoterics (HIPE).

"Our ability to measure many small, detailed aspects of our universe is far ahead of an ability to make meaningful use of the data.

Continued next page ...

"It appears to me that the real value of TEF analysis will become apparent when we more fully understand what matters and what doesn't.

"I appreciate your publishing Mr. Clark's comments. They certainly help me to be more confident of my own observations."

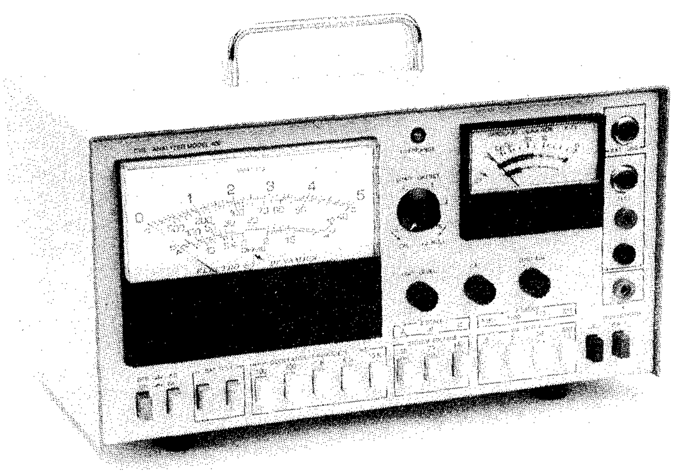
Additional points from Jack Wrightson:

"A recent re-reading of early articles concerning the LEDE concept reminded me of another example of one of the points I was attempting to make in my last letter.

"In one article, it was stated that an anechoic studio control room was undesireable. This may be true, but not for the reason mentioned. It was stated that the lack of any diffuse field would produce unsatisfactory tonal balance for a listener anywhere other than the optimal mix center due to comb filtering. As seen on an instrument (i.e., TEF gear) there is a gross spectral aberration; the ear, however, typically interprets the time delay between loudspeakers as a shift in location of the sound. It is important to remember that our hearing mechanism is most adept at interpreting signals for their behavioral (environmental) value. It is well known with experiments utilizing headphones that listeners can be trained to hear a particular stimuli as either a pitch shift or a change in perceived location. In other words, we can train ourselves to trigger different ear/brain interpretations for the same signal. Virtually all land vertebrates can move their hearing organs to better localize sounds. As we all know, when humans do this, there is little or no change in the quality of the sound.

"The danger is, as Mr. Clark mentioned, our instruments inform about the nature of the signal reaching, say, a microphone. They do not tell us what signal reaches our two ears or how our ear/brain system interprets these signals. The relative interaction and independence between the two ears is a topic that contains many unknowns and much active research."

ELECTRO-DYNAMICS IMPEDANCE METER



We show the Electro-Dynamics Impedance Meter in many of our classes and recommend it.

From Wayne Pommer, President of Electro-Dynamics Development Co., we received the following comments:

"I have made a few changes in the Model 400 in the past few months, all to the betterment of the instrument. The range and internal calibration resistors have been tightened from 1% to .1%. The PC board layouts have been changed and the Molex 3-pin connector is now 6-pin to facilitate servicing. Also, there is now more precise calibration of auxiliary functions--oscillator level metering and battery test.

"I am offering a multi-end test lead kit (at the suggestion of Elaine Schiller) and am playing around right now with a rechargeable battery pack, though that is in the preliminary stages of development."

Wayne refers to ELAINE SCHILLER. Elaine, seen here talking to Don Eger, has attended several classes, including the first Heyser TDS class. Her title is Head, Speech Systems & Interference Branch, Naval Ocean Sciences Center, in San Diego.

Electro-Dynamics address is 475 Mt. Hood Dr. S.W. Issaquah, WA. 98027.

BOOST VS. CUT

EDITOR'S NOTE: On occasion Syn-Aud-Con graduates rise to rare heights of technical discourse. TED UZZLE'S explanation of when and where to use boost and cut filters reminds me of a famous acoustician's remark upon being asked by an architect, *"Dr. ______, where shall I place the sound column?"* To which the good Dr. said, in a stage whisper to those of us witnessing the request, *"You know, I almost told him."* The application note, authored by Ted, and reproduced here, finally finds a use for a boost filter just as there surely is one for sound columns. If you don't have one of the problems listed in this application note, *don't use a boost filter.*

BOOST VS. CUT by Ted Uzzle

For many years, Altec Lansing manufactured cut-only equalizing filter sets and advocated the use of this type of device in correcting loudspeaker/room frequency response. In 1981 Altec began production of filter sets that both boost and cut in each band. This Applications Note attempts to explain the relative advantages and disadvantages of boosting and attenuating filters for the many applications to which they are put.

An equalizer is any device intended to change the frequency response of an audio system; they have been used in circuitry for band-limiting since the beginning of electrical recording and amplification. About 45 years ago John Volkman of RCA equalized a motion-picture theatre playback system, but the equipment and instrumentation of the time were so cumbersome that no further use resulted. The first technically and commercially successful acoustic equalization was by Wayne Rudmose in Love Field in Dallas, and described by him in a paper in *Noise Control* for July 1958.

The earliest graphic equalizing filter sets were built by A.C. Davis' Cinema Engineering Co., Westrex, Langevin, Altec Lansing, and others for the motion picture industry many years ago. These units usually had bands of about 1½ octaves each, and would both boost and cut in each band (they called the boost side "equalize" and the cut side "attenuate"). They were used for post-production processing and sweetening, and to this day you will find many of these hardy pioneers installed smack in the middle of modern mixing consoles in the Hollywood studios.

Units such as the Altec 9062 graphic equalizing filter set were not intended to adjust the acoustic frequency response of playback systems, nor to prevent feedback (although the careful use of one by Robert Ancha in a Chicago hotel ballroom in 1967 allowed a successful equalization). Their sole purpose was the creative and artistic control of a recording, and also mitigation of signal-to-noise ratio problems in location recordings. It was not until years after the 9062 was introduced that C. P. Boner would exploit the fact that when feedback occurs, it does so at one discrete frequency, which can then be eliminated with a very narrow notch filter, usually about 50 Hz wide at the -3 dB points. This was the essence of the concept: reject the feedback frequency with a cut-only circuit.

When the original Altec Acousta-Voice system was developed in 1967 by A.C. Davis, D.B. Davis, and J. J. Noble, it continued the idea of rejecting those areas of the spectrum with a likelihood of feeding back, and accordingly was an attenuate-only unit. By using contiguous, critical bands, Acousta-Voicing made feedback equalization easy, fast, stable and reliable. It did much more, however; in fact, improvement in feedback behavior was only number 15 of the 18 claims in the Acousta-Voice patent. The major object of Acousta-Voicing was to provide an improved reinforcement or playback system, particularly in association with the acoustics of the room in which the system operates. Thus was born the idea of tailoring the frequency response of the sound system/room combination.

There were still good reasons for Acousta-Voicing to be an attenuate-only equalizer.

An acoustic frequency response curve can be characterized as a series of humps and dips around some hypothetical center line. It is important to understand that all humps can be cut, as deeply as need be, with safety to the sound equipment and without violation of the basic gain structure of the system.

A hump can be caused by the directivity of the loudspeaker. A 4 dB hump in a certain band may indicate no more than a Q value rising by a factor of 2.5: this is simply a function of directivity index. Lowering the random-incidence pressure response in this band will alter the total power radiated by the loudspeaker, but the direct to reverberant ratio will be maintained in the given band and the ear will usually prefer the improved spectral balance.

Another acoustic characteristic creating a hump can be a deficiency in absorption in the room. In the case of large rooms with almost no low frequency absorption, the resulting boomy sound can be quite damaging both to intelligibility and to naturalness, and reducing the power output of the loudspeaker at low frequencies can improve both.

The only response peak whose elimination is not an improvement is one creating a rattle. The author first encountered this in a motion picture theatre whose sound had a guttural, most unpleasant quality. After considerable work on the system and acoustic response, without improvement, it was discovered that there were wooden folding chairs stacked shoulder-high behind the screen, and that one frequency set them rattling at very high levels. Swept sine-wave rattle tests are as good an idea today as they were years ago, but the practice has fallen into general disuse.

Continued next page ...

While humps can always be eliminated by equalization, and this is almost always an improvement, few dips can be boosted to the benefit of the overall sound quality and system stability. The range limitations of transducers can seldom be widened by equalization in the electronics; in general, a need for much more woof and tweet can be met only by more, or different, transducers.

Cancellation at crossover is a classic example of a problem equalization cannot cure. If one driver is connected with reversed polarity, or the measurement microphone happens to be situated right between lobes in the combined pattern of the high and low frequency devices, absolutely no degree of boost will restore the dips.

Another classification of dip not amenable to equalization is created by selective absorption, which slices out a band of the sound energy in the room and leaves the broader spectrum untouched. A Helmholtz or slot absorber may have been introduced deliberately for acoustical purposes, or may have been created quite by accident. A panel free to resonate is usually accidental. In either case, no amount of boost will restore the response properly, while large amounts of boost will introduce a variety of other problems into the system.

The unloading of a horn or low frequency saturation of a transformer somewhere in the sound system are two more examples of dips in response equalization cannot cure.

Most attempts to solve acoustic problems by boosting will degrade overall system listenability. It is a grotesque irony, often met in systems with passive crossovers, that an attempt to extend the low frequency response of a loudspeaker system by pushing hard with equalization will result in blown high-frequency diaphragms! The proximate cause, of course, is clipping in an overdriven amplifier.

Further, inexpertly designed boost and cut filter sets can introduce enormous phase shifts, sometimes discontinuities, when adjacent bands are alternately boosted and cut. Also, where the signal-to-noise ratio is thin, a boost in a given band maintains the signal-to-noise ratio in that band but degrades it overall. Finally, careless boosting in weak regions of the spectrum will often destroy transducers.

These differences between boost and cut are to be expected when the nature of acoustic interaction is considered. Two acoustic sources interacting physically can only add to a specific level, but they can cut (as for instance by phase cancellation) to any depth, including the entire loss of a signal. In acoustic equalization, the weakest link breaks the chain; relaxing the strong link wastes its capacity, but straining the weak one can destroy everything.

This is not to say that boost equalization has no value. Often it will be the only practical way to achieve a set of pre-emphasis/deemphasis curves that will maximize signal-to-noise ratio through a noisy link. It is obviously of use in the new amplified musical styles, where naturalness is not considered essential. It is appropriate for special effects, for example in conjunction with a subwoofer or supertweeter whose purpose is to impress the listener (and not suspend his disbelief in the artificial reproduction of sound that is taking place.)

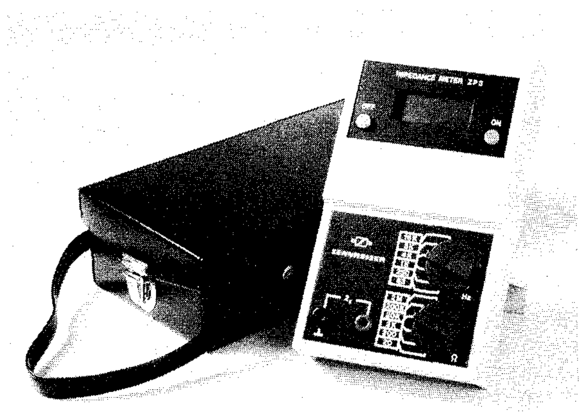
Mechanical faults and limitations usually require mechanical solutions: in the mechanical domain equalization is not useful. In the electrical domain both the boost and cut segments of a minimum-phase filter set can be used to correct a minimum-phase amplitude response perturbation, and the phase response will also be corrected to flat. In the domain of acoustic behaviors, discretion demands that cut be used primarily. In order to use an equalizer properly the engineer must know his purposes clearly: understand the *physical* meanings of the various humps and dips he sees on the display of his real-time analyzer; respect the fact that some cannot be corrected electronically; and always remember that equalization is not a magic touch capable of curing scrufola.

MARY GRUSZKA

One of our most avid, eager "learners" is MARY GRUSZKA, engineer at CBS-TV in New York. Mary attended her first of many Syn-Aud-Con classes while a student at Syracuse University in 1974. In 1982/83, Mary attended the Concert Hall Workshop, TEF® and Loudspeaker Array Workshops.

SENNHEISER ISSUES NEW IMPEDANCE METER

Many of us used the old Sennheiser impedance meter with satisfaction and the appearance of its replacement, the new ZP-3, is a welcome addition to available instruments.



Sennheiser ZP-3 Impedance Meter

The unit measures impedance from 1.000Ω to 1.999 MΩ in six ranges at frequencies of 63, 250, 1000, 4000, 8000, and 16,000 Hz. Accuracy is ± 5% and readout is via a large, easy-to-read 3-1/2 digit liquid-crystal display. Measurement voltage is very low (≤ 85 mV) enabling safe measurement of tape recorder heads, microphones, transformers, musical instrument pickups, vibration sensors, etc., without damage to windings or risk of magnetization as might occur with an ohm meter.

When measuring *simple* R-C, R-L or L-C networks or combinations thereof, the availability of six measuring frequencies enables "untangling" the separate component values from the complex network via observation of the network's impedance-vs-frequency behavior.

Note the italicized *"simple."* Measuring a long telephone line with "distributed parameters" rather than "lumped parameters" does not allow this simple solution.

Additionally, the ZP-3 may be used as a convenient portable test-tone generator, providing approximately an 85 mV (-22 dBV) output. (Exact output dependent on the input impedance to which the ZP-3 is connected.)

The ZP-3 is powered by two standard 9 V transistor radio batteries and automatically shuts off after 3 minutes of operation. ("LoBatt" displayed in readout when batteries need to be replaced.)

Additions to the Unit We'd Like to See

We'd prefer that there was a jack for an external oscillator so we could check other than the built-in frequencies. We'd very much like (probably a matching companion unit) to have it read out the phase angle along with a simple indication of whether the voltage or the current was leading (perhaps an LED that lit for "ELI" in "ELI" the "ICE" man).

In our checks of the unit against our Grützmacher bridge we found it to be accurate and rapid to read. Sennheiser misdefines impedance in their catalog as "The impedance of a device is the AC resistance measurable at the terminals." (Page 6 of their current catalog) Fortunately, the ZP-3 measures the magnitude of the complex impedance not the ACR. Actually, we'd like a switch on the ZP-3 that allowed us to read both Z and ACR since with that data the phase angle is immediately available.

$$\text{Phase angle} = \arctan\left(\frac{\sqrt{Z^2 - ACR^2}}{ACR}\right)$$

The ZP-3 is useful as it is. With both Z and ACR plus an external oscillator, it would be an outstanding buy. Suggested professional user price is $530. For more information, contact Sennheiser Electronic Corporation (N.Y.), 10 West 37th Street, New York, NY 10018.

Those of you already thoroughly familiar with Laplace transforms, Hurwitz filters, s planes, poles and zeros are fortunate indeed. You'll be among the first to come to a full realization of the utility of the Crown TEF® analyzer's "Nyquist" phase plots (relative phase) and their representation of non-minimum phase frequencies as an epicycle. For those of you not familiar with these remarkable tools, a review of Syn-Aud-Con Tech Topic Vol. 5 No. 10 by Gerald Stanley is a must.

We are on the threshold, at long last, of viewing economically and efficiently, with high resolution, the full temporal behavior of the loudspeaker *systems* we use. Now is not too early to begin serious study of the domain chart in Tech Topic Vol. 8 No. 5, p. 4. Those measurement parameters not currently familiar to you should prompt research into the better articles in the literature.

We have recently had the privilege of having a number of graduates introduce us to a remarkable resource. Douglas Preis of Tufts University, Medford, Mass., has published some truly relevant tutorial material on "Linear Distribution" JAES Vol. 24 No. 5, June 1976 and "Phase Distortion and Phase Equalization in Audio Signal Processing -- A Tutorial Review", an AES preprint 1849 (J2) Oct. 1981. This material answers myriad questions and guides the reader into a solid conceptual feeling for some of the parameters we are soon going to be exploring in detail.

Phoenix - 1983

San Francisco - 1983

LOFTECH TS-1 AUDIO TEST SET

If one is willing to overlook being addressed "Dear Conslultant" and being told in the instruction manual that there are four zero VU levels in the audio industry, and we are willing, then you should look into LOFT professional audio products. The unit we have been testing is their Loftech TS-1 Audio Test Set.

What Did We Like?

It's a super lightweight unit (4.4 lbs.) that can easily fit in any attache case. It consists of an extremely useful continuous sweep from 20 to 20,000 Hz oscillator, a digital frequency meter, and an extremely wide range relative dB meter (-50 to +24 range), referenced to 0.775V, with a back of the unit sensitivity control that allows you to turn it into a dBV meter (dB referenced 1.0 volt open circuit). It arrives calibrated to read 0.775 volts = 0 dB essentially open circuit as the output Z = 50Ω. The input Z of the meter is 100,000Ω.

Loftech meets or exceeds all their specifications and our experience with the unit is that it always works and is easy to use. It's rated at +10 dBm output. Actually, what they mean is that placed across a 600Ω resistor, the voltage amplitude is +10 dB. To find the available input power, AIP:

$$AIP = 10\ LOG\left(\frac{(E_S)^2}{0.001\ R_S}\right) - 6.02\ dB$$

If R_S = 50Ω and we are connected to 600Ω, then: $\qquad E_S = E_{IN}\left(\frac{R_S + R_{IN}}{R_{IN}}\right)$

$$E_S = 0.775\left\{10^{\left(\frac{10}{20}\right)}\right\}\left(\frac{50 + 600}{600}\right) = 2.65V \quad \text{and:} \quad AIP = 10\ LOG\left(\frac{(2.65)^2}{0.001(50)}\right) - 6.02\ dB = 15.47\ dBm$$

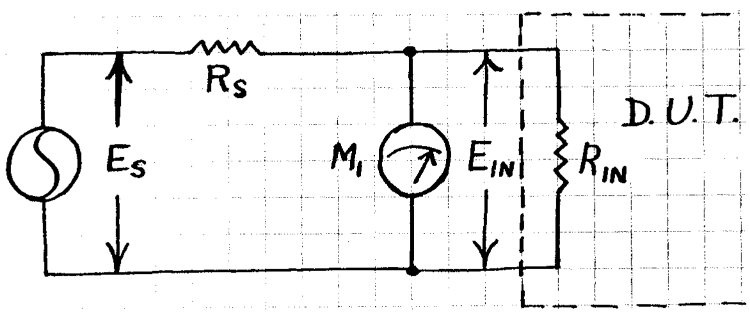
What Loftech calculated was not the AIP but the output level across 600Ω as a load (R_L).

$$\text{Power across a load in dBm} = 20\ LOG\left(\frac{E_X}{0.775V}\right) + 10\ LOG\left(\frac{600\Omega}{R_L}\right)$$

If R_L = 600Ω, then: $\quad E_X = 0.775\left\{10^{\left(\frac{10}{20}\right)}\right\} = 2.45V \quad$ and $\quad 20\ LOG\left(\frac{2.45V}{0.775V}\right) + 10\ LOG\left(\frac{600}{600}\right) = +10\ dBm$

We found their instruction book useful in terms of describing physical hookups but it should have pages 5 through 10 torn out as well as page 21. Pages 75 and 76 describe a useful technique for impedance measurements of inputs of devices the unit is driving. The following drawing and equations explain the theory and also reveal why using your calculator would be better than using their table.

COMPUTING APPROXIMATE IMPEDANCE OF A DEVICE'S IMPUT CIRCUIT FROM A CHANGE IN A VOLTMETER'S AMPLITUDE (EXPRESSED AS A RELATIVE LEVEL IN dB REF. TO 0.775V) PLACED ACROSS THE IMPUT TERMINALS



M_1 has a Z = 100 KΩ

R_S = 50Ω

THEORY

$$\frac{E_S}{E_{IN}} = 1 + \left(\frac{R_S}{R_{IN}}\right) \equiv 10^{\left(\frac{dB}{20}\right)} = 1 + \left(\frac{R_S}{R_{IN}}\right)$$

$$\text{Thus:} \quad R_{IN} = \frac{R_S}{\left(10^{\left(\frac{dB}{20}\right)}\right) - 1}$$

Where: The difference in relative level is expressed as a positive number

$$\text{And:} \quad N\ dB = 20\ LOG\left(\frac{R_{IN}}{R_S + R_{IN}}\right)$$

EXAMPLE

R_S = 50Ω $\qquad$ meter indication is -0.7 dB

Input Z = ?

$$R_{IN} = \left(\frac{50}{\left[\frac{1}{10^{\left(\frac{-0.7\ dB}{20}\right)}}\right]^{*} - 1}\right) = 595.76\Omega \quad \text{and} \quad 20\ LOG\left(\frac{595.76}{50 + 595.76}\right) = -0.70\ dB$$

Continued next page....

*If the indicated dB change is negative, you can substitute:

$$\left(\frac{1}{\left(10^{\left(\frac{-dB}{20}\right)}\right) - 1}\right) \quad \text{for} \quad \left(\frac{1}{\left(10^{\left(\frac{+dB}{20}\right)}\right) - 1}\right)$$

$$E_X = 0.775V\left(10^{\left(\frac{\pm dB}{20}\right)}\right)$$

For Syn-Aud-Con graduates presently without an oscillator, frequency meter and some way of reading *relative* levels, we are pleased to give this product our highest approval as an eminently practical solution to your problem. Best news of all -- it's only $299.00.

Loft Professional Audio Products are manufactured by Phoenix Audio Laboratory, Inc., 91 Elm Street, Manchester, Connecticut 06040 - (203)649-1199.

A "COVERAGE" APPROACH TO ARRAY DESIGN

With the advent of really useful, highly accurate array mapping with JOHN PROHS - Community's CLUSTER COMPUTER, it is possible to begin an array design by first considering coverage. The consequences of taking this choice are:

1. "N" is determined initially by coverage considerations rather than intelligibility considerations.
2. "$Q_{MIN}(ss)$" can be determined in the normal way for each device in the array and then multiplied by "N".
3. A "$Q_{(AVAIL)}$ may be chosen and the relationship $N = \frac{Q_{MIN}(ss)}{Q_{AVAIL}}$ must be $\leqq N_{COVERAGE}$ followed by: $D_{2MAX} = \frac{D_2ss}{N}$ if required.

Notes on the Above

We normally suggest that in designing arrays, you assign L_W only to individual devices and account for the array's L_W by using "N". For example, for any given device's L_W, the total L_W of the array then becomes:

$$L_{W(TOTAL)} = L_{W(DEVICE)} + 10 \, LOGN$$

When the "N" term is used in association with "Q" the proper form is: $NQ(\;)$

Problems to Avoid in Array Design

1. Assuming that a reverberant sound field (L_R) exists when, in fact, it does not.
2. Using the L_W of an array to calculate the L_D of a device at a given observation point.
3. Attempting to assign the array a Q value, rather than calculating the N factor.
4. Using a device's L_W to calculate L_R rather than the total L_W.

$$L_{W(TOTAL)} = L_{W(DEVICE)} + 10 \, LOGN$$

5. Assuming that Q is for an area instead of a point.
6. Misunderstanding that the changing D_C with changing Q for differing loudspeaker angles is accompanied by *changing levels*, as well, when calculating acoustic gain and consequently trying to account for it in your calculations.

Conclusion

Array design requires an orderly process in order to account for the interacting variables of Q, N, D_2, $\bar{Sa}$, as they interface the complex sound fields present in a real environment.

Always be alert for the totally unexpected psychoacoustic serendipity that can on occasion occur in spite of the horrendous measurement staring you in the face - just be careful of auto-suggestion.

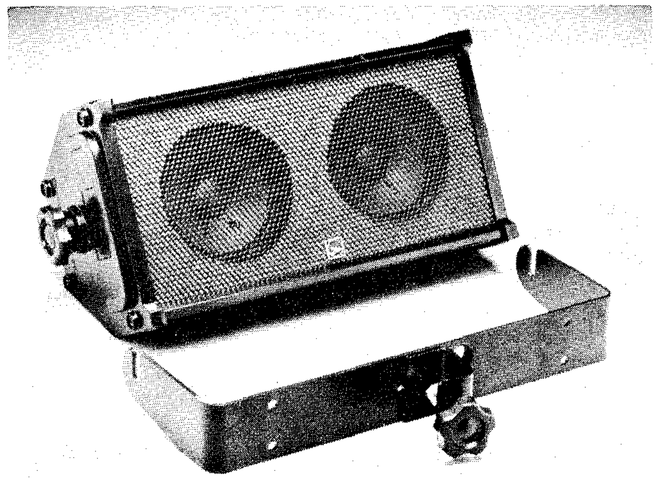
NEW SYN-AUD-CON MEASUREMENT LOUDSPEAKERS

Syn-Aud-Con likes to use small loudspeakers for demonstration purposes during classes. We have tried a variety of such units, many of which had excellent amplitude versus frequency curves but often were not very rugged.

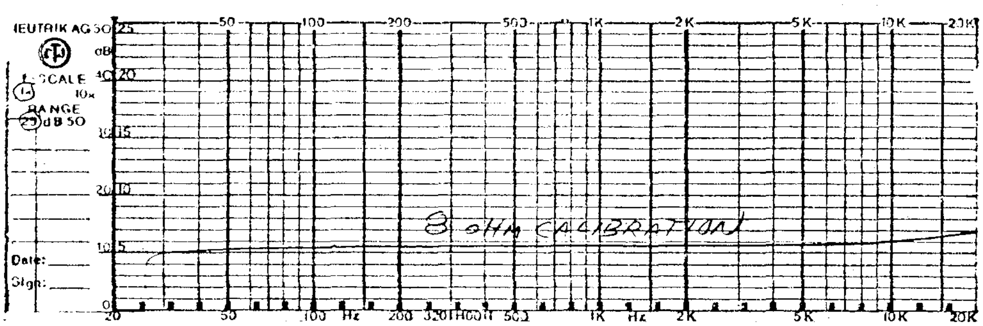
We recently used a pair of TOA RS-21M units. The tests shown here were made on our new TEF®* analyzer (Tecron System 10). These are small 6.7" high by 5.9" deep by 6.7" wide and weigh only 5.3 pounds. They are rugged enough to allow us to make "standing wave" demonstrations in the 70 to 125 Hz region during classes. We were pleased to see that our measurements matched the measurements published on their data sheets that come with these units (after linear frequency scaling is converted to log frequency scaling).

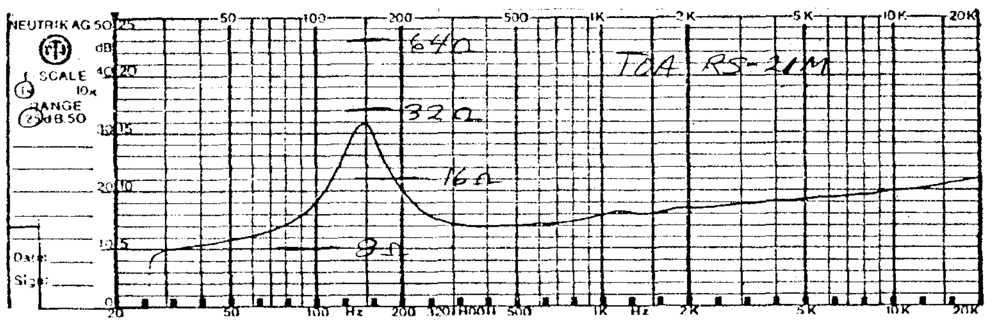
A second TOA unit we tested was their SM-60. This is a dual speaker system using the same 5" driver employed in the RS-21M units. It, of course, exhibits the same difficulties all dual speaker systems have but where high power handling capability in a small size compensates for danger of comb filters, this very rugged unit with its versatile mounting hardware, cast aluminum housing and locking phone plug connectors is an ideal choice.

TOA RS-21M.



TOA SM-60.





*Trademark of Crown/Tecron

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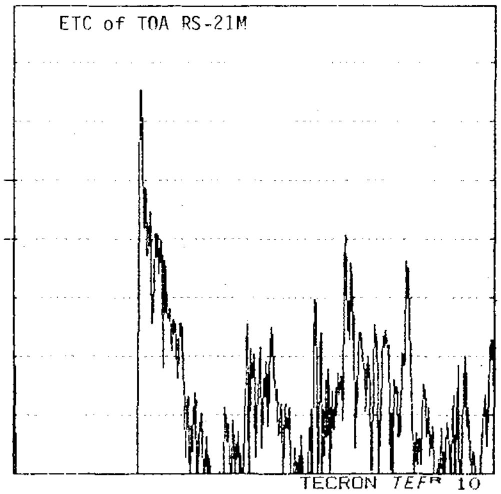
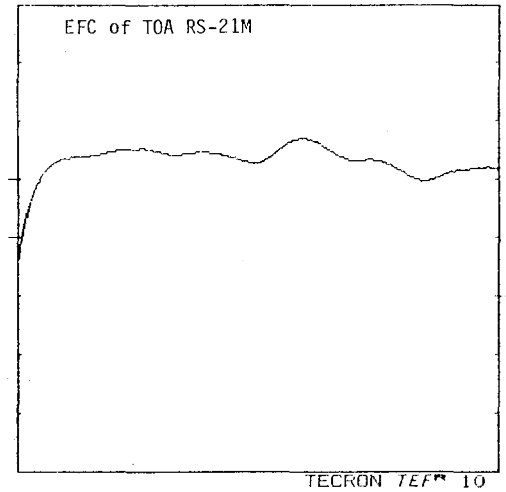


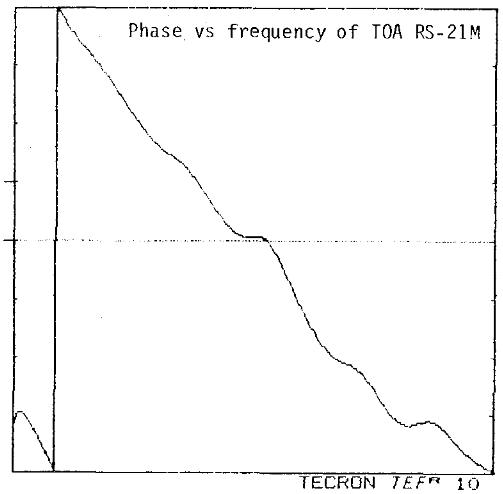
Vertical: 6dB/div with base of display at 54.3dB
0dB is located at .00002 Pascals

Horizontal: 0 microseconds or 0 Feet to
12889 microseconds or 14.565 Feet
scale: 3.9822E+00 Feet/inch or 1.5678E+00
Feet/cm.



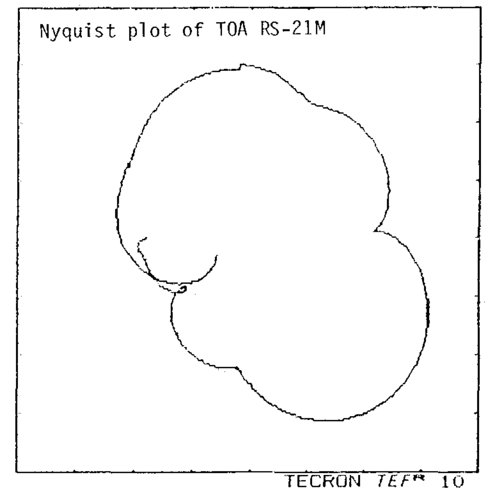
Vertical: 12dB/div with base of display at 30.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 9868.43Hz
scale: 2698.12Hz/inch or 1062.25Hz/cm.

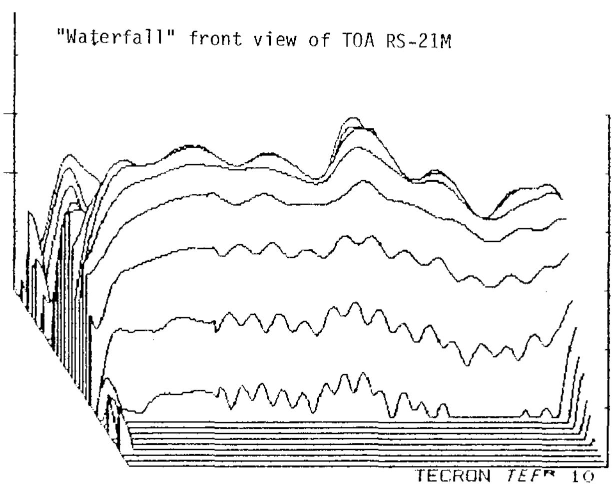


Vertical: 45 degrees/div.
0 degrees is at the dotted horizontal line.

Horizontal: Auto 0.00Hz to 9868.43Hz
scale: 2698.12Hz/inch or 1062.25Hz/cm.

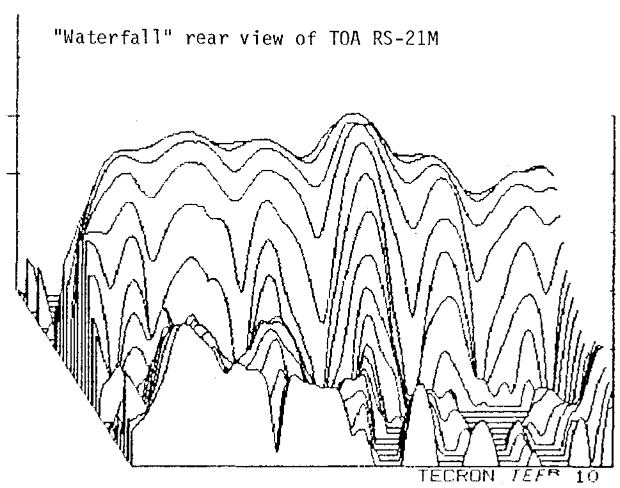


Resolution: 1.0527E+00 Feet & 1.0735E+03Hz



Vertical: 6dB/div with base of display at 72.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 9868.43Hz
scale: 2698.12Hz/inch or 1062.25Hz/cm.



Vertical: 6dB/div with base of display at 72.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 9868.43Hz
scale: 2698.12Hz/inch or 1062.25Hz/cm.

THE PSYCHOLOGY OF MUSIC

This brand new, important, exciting, and authoritative book is edited by Diana Deutsch of the "Deutsch Effect" we demonstrated in classes last fall. Mrs. Deutsch was our guest at the seminar center this winter and we were impressed not only with her depth of background experience, but with her genuine creative insights into new and important ways to explore psychoacoustic problems.

Diana Deutsch talking with GARROTT ELGHAMMER during a TEF® Workshop

Paul D. Lehrman has written an unusually perceptive review of this book for REP Magazine (partially quoted below.)

"To be in the recording business today, you have to be at least somewhat aware of the science of acoustics - the way that sound behaves in a physical environment. But chances are likely that you've never dealt much with the field of psychoacoustics - the way that sound behaves once it reaches the ear, and starts to get processed by the brain. Of course, a few psychoacoustic principles are applied, consciously or not, in the day-to-day work of recording - like the Fletcher-Munson curve, which says that the ear is less sensitive to tones at the low and high end of the spectrum at low volumes. Also, the fields of studio and equipment design use some of these ideas - such as LEDE rooms and, in the world of consumer hi-fi, the Carver 'Sonic Holograph,' both of which take advantage of recent research in phase sensitivity.

But going beyond these rudimentary applications is a wealth of research that has been done by scientists, far removed from the recording industry, on the way the brain-ear combination actually hears sound. Many in our field are familiar with 'On the Sensations of Tone,' by the 19th-centruy physicist Hermann von Helmholtz. At the time, the book was the most comprehensive look at the subject, and it is still a very useful reference for many applications. But it was written 120 years ago, and the field of psychoacoustic research has travelled a long way since then. Helmholtz, for one thing, was severely limited in his work by the fact that pure tones for his experiments simply were not available. He had to rely on musical instruments for his sound sources which, as we know, produce tones that constantly change--even if two trumpet tones sound the same, a quick check with an oscilloscope can verify that their absolute pitch, attack, timbre, and envelope may vary widely. Given these restrictions, it's not surprising that many of Helmholtz's conclusions were misleading, or even downright wrong.

It's only since development this century of electrical sound synthesis that psychoacoustics has progressed into an exact science, and more recent applications of computer technology have resulted in further refinements. That is the point made in the opening chapter of this new book edited by Diana Deutsch, a psychologist at the University of California, San Diego-La Jolla. Deutsch has collected 18 essays on various aspects of the field, written over the last decade by a wide spectrum of European and American contributors: professors of psychology, music, and audiology; researchers in acoustics, electronics, and perception; and musicians.

Each chapter is of value for anyone involved in the recording industry, but some hold extra interest for specialists in certain fields. For example, Chapter 2 - 'Exploration of Timbre by Analysis and Synthesis' - makes fascinating reading for anyone working with synthesizers, particularly if they are attempting to simulate real musical instruments. It shows how the synthesis of natural tones is an incredibly complex process, but then offers shortcuts that effectively fool the ear. Chapter 3, 'Perception of Singing,' is of at least as much interest to singers as it is to psychologists, and helps to explain some of the difficulties in recording singers in a natural-sounding environment. Chapter 4, 'Grouping Mechanisms in Music,' is full of information for composers and arrangers. It explains, for example, how the ear often takes widely disparate notes and turns them into melodies that aren't really there. Chapter 7, 'Timing by Skilled Musicians,' makes the startling conclusion that even the finest musician's sense of rhythm is often way off--and explains why some people feel that drum machines sound somehow inhuman.

Other chapters deal with absolute pitch (how do we learn it, and what is it good for?); the role of interaction with other musicians and hearing mechanisms in musical performances (monitor engineers take note); the limits of musical memory; the role of music in social situations (does the Rite of Spring *cause traffic accidents?); and the mathematics and psychology of musical scales."*

One illustration from the book indicates the kind of vital, yet hard-to-obtain research data it contains:

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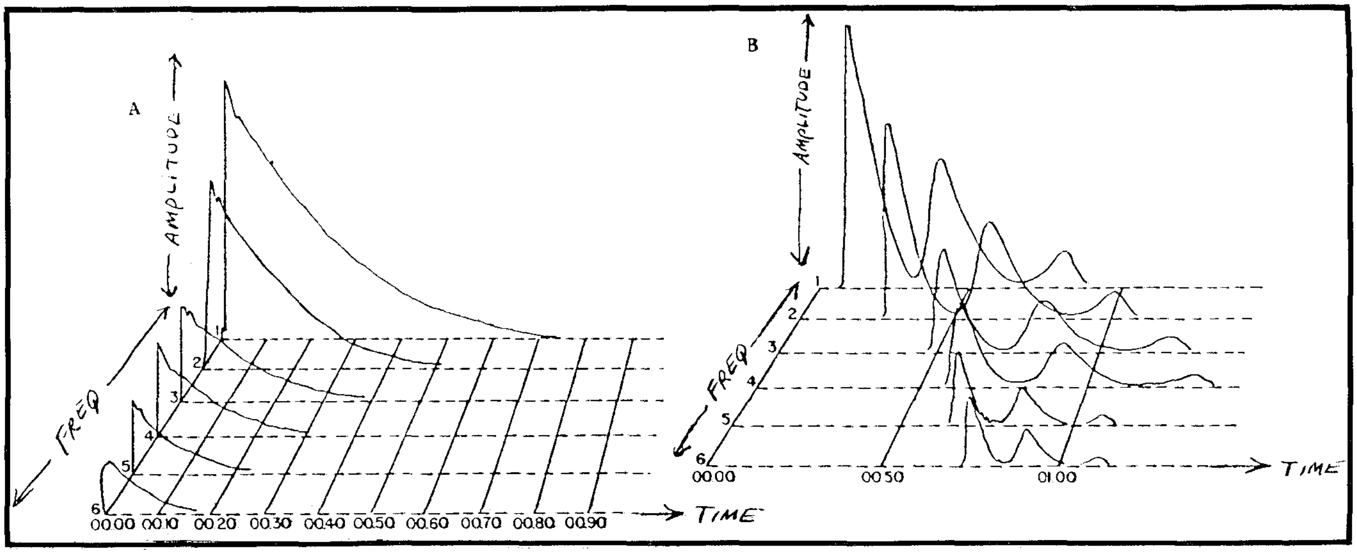

Perspective plots of synthetic inharmonic tones: the vertical axis is amplitude, the horizontal axis is time, and the depth axis is frequency. In A, the sharp attack followed by a decay yields a bell-like tone. In B, the time-varying amplitude function yields a fluid nonpercussive tone, in which the components can be heard much better than in the fused bell-like tone. (In spite of the appearance of equal spacing these tones have nonharmonic components.)

We've seen popular loudspeakers with serious "comb filtering." Are they liked for their reproduction of components within the complex whole? If so, how does the decay side of the 3D response behave? Does it change time with frequency? Fascinating, isn't it?

THE PATH OF LEAST RESISTANCE

During our annual (whenever we're on the road) visit to the STEVE SIMPSONS in San Antonio, Texas, we noticed the message reproduced here framed and hanging in the foyer of their offices.

While generally inclined to walk away from conflict if allowed to do so, I recognize that much that I respect in the world is preserved by quiet but fierce fighters. It's axiomatic that those who achieve nothing are usually free of criticism. In the situations where there are no second place winners, the path of least resistance leads to slavery or the grave. Free men are fighters willing to pay the price freedom costs without fear of the the consequences. George Bernard Shaw said it this way:

> "The reasonable man attempts to conform to the world as it is.
>
> "The unreasonable man attempts to make the world conform to him.
>
> "Therefore, all progress is made by unreasonable men."

THE PATH OF LEAST RESISTANCE

The path of least resistance is
A smooth and easy one.
It's wide and has no hills to climb
Or bumpy spots to shun

It's like a lovely fishing stream;
It always flows down hill.
It never seeks the mountain's top
Where views will always thrill

The path of least resistance goes
A long and quiet way
And those who travel it are safe
For they will see no fray.

But there are those who dare to climb
Up paths not always clear
And battle for the great rewards
Not won through ease or fear

Those paths are often rough and steep
And some get hurt or lost
But those who make it win the prize
And don't regret the cost.

They turn the wheels: they thrill the world
And all mankind is blest
They sought the top: would not coast down
As do some of the rest

The paths of least resistance are
An easy way to go
But victors seek the mountain's top
Not known to those below.

QUOTES

Sit down before fact like a little child, and be prepared to give up every preconceived notion, follow humbly wherever and to whatever abyss Nature leads, or you shall learn nothing.

T. H. Huxley

For the laymen whose common sense is so rudely trammeled by the modern point of view, (In Physics) perhaps there is consolation that these bizarre notions are shocking and abrasive to physicists themselves. Their capacity to handily integrate new ideas is, in fact, limited, and has given rise to the observation that *physicists never really understand a new theory, they just get used to it.* (Underlining mine).

Dr. Larry Dossey

A human being is part of the whole, called by us 'Universe', a part limited in time and space. He experiences himself, his thoughts and feelings, as something separated from the rest - a kind of optical delusion of his consciousness. This delusion is a kind of prison for us, restricting us to our own personal desires and to affection for a few persons nearest to us. Our task must be to free ourselves from this prison by widening our circle of compassion to embrace all living creatures and their whole nature in its beauty. Nobody is able to achieve this completely, but the striving for such achievement is in itself a part of the liberation and a foundation for inner security.

Albert Einstein (Letters)

Anyone who cannot cope with mathematics is not fully human. At best, he is a tolerable subhuman who has learned to wear shoes, bathe, and not make messes in the house.

Robert Heinlein
(Notebooks of Lazarus Long)

Most of us think of mathematics as those chicken tracks - little wiggle signs. That isn't math! - that's the fossil remnants of a thought.

The thought is the math. It is the structured reasoning that is the math. And when you start taking things we refer to as common sense and observation and you begin to structure that is a reasoning mode - that's math.

The axioms and postulates of that which most of us would call common sense - that's math! When it's dried up and withered and appears as little chicken tracks on a piece of paper - that ain't math! - that's just the residue of it - just a shorthand that lets people know that a mind went past here on this page.

Math is structured reasoning.

Richard C. Heyser
(Special Graduate Seminar)

ARTICLES OF INTEREST

A remarkable "tutorial paper" in the June 1982 JASA, pp. 1321-1334, entitled "Auditory Perception of Radio-Frequency Electromagnetic Fields" written by Chow and Guy, leads one to realize that the human being is indeed constructed in mysterious and marvelous ways.

> *Absorption of pulsed microwave energy can produce an auditory sensation in human beings with normal hearing. This phenomenon manifests itself as a clicking, buzzing, or hissing sound depending on the modulating characteristics of the microwaves... That is, one hears sound because a miniscule wave of pressure is set up within the head and is detected at the cochlea when the absorbed microwave pulse is converted to thermal energy.*

CLASSIFIED

Continued next page.....

CLASSIFIED continued

FOR SALE:
- Complete White Instruments 400 plug in series equalization filter set. Contains six broad band filters, 23 1/3 octave filters, 100 narrow band filters and custom carrying case. Retail - $16,610.00.....Will sell for...$4,750.00

  Contact: Don, Sound Investment Enterprises, P. O. Box 4139, Thousand Oaks, CA 91359 (602)585-4093 or (213)991-3400.

FOR SALE:
- Syn-Aud-Con's Gen Rad 2512 FFT Analyzer, Serial #169. A new version of this unit sells for $18,000. +. Our unit, like new, is . . . . $9,500.00 Complete manual and carrying case (anchor) included.

FOR SALE:
- HP 333A Auto-Null Distortion Analyzer .... $ 995.00
- Crown IMA Intermod Anal .... $ 350.00
- Pair of custom-built University "Classic" S9S three way horn-loaded enclosures with walnut veneer and extra 15" woofer, freight extra........ $ 995.00

  WANTED: Instructions or schematic for Source Engineering Noise Filter.

Contact: CHRIS HOOD, P. O. Box 44110, Crafton Borough, Pittsburgh, PA 15205. Phone: (412) 921-2911

FOR SALE:
- Shure MG15A5 Audio EQ System .... $ 175.00
- IVIE IE-10A Analyzer .... $ 250.00

Contact: Tim Kelley, Rt. 2, Box 239C, Braham, MN 55006. Phone: (612) 396-2832.

FOR SALE:
- 2LV with PX-TL Power Supply
- 30 GPB with PX-18 Power Supply
- 6 LPB with PX-18 Power Supply ....Under Warranty.................... Make offer

Contact: Nancy Dykstra, 1301 Brocton Lane, Charlotte, No. Carolina 28211. Phone: (707) 365-0411

FOR SALE:
- We found a box of 24 Hearing Protectors in storage. These are David Clark 27L Hearing Protectors with the Syn-Aud-Con logo on them. 24 is not enough to list on our order form, so we have listed them here. $ 18.00 each plus $ 2.00 handling

Contact: Syn-Aud-Con, P. O. Box 669, San Juan Capistrano, CA 92693. (714) 496-9599

FOR SALE OR ASSUME LEASE:

- Hewlett Packard #85 microcomputer *
- Hewlett Packard #82901 Flexible Disc Drive
- Microsystems #9800 Winchester Drive (5 megabytes)
- Anadex #DP 9501 Printer

*Includes: #82939A Serial Interface
#82937A (HP-IB) Interface
#82903A 16K Memory Module
#85-15003 Input/Output
#85-15004 Matrix
#85-15005 Advanced Programming

We would be willing just to have someone else assume the lease payments with no need for any payment to us for what we have paid. The computer and its software are on two different leases with the same company (Equico Lessors) and are detailed below:
(a) Payment of $83.71 monthly until March 1987; (b) Payment of $211.22 monthly until September 1985..

Contact: A. Robert Jack, D.V.M., Villa Animal Hospital, 4250 East Chapman Avenue, Orange, CA. 92669. Phone: (714) 633-9780.

EMPLOYMENT WANTED:

Syn-Aud-Con Graduate, 1976. Updates: 1978, 1982. Member of AES-15 years. AF of M-14 years.
Over 20 years experience including:
- Sound System Design, Fabrication, Installation.
- On location recording - Symphonic, Band, Soloists, etc.
- Live Reinforcement - Stage shows, Concerts, Presidential visit.
- Industrial - Electronic & Electrical Maint., Test Jig Design.
- Supervision - Quality Control, Semi Conductor Mfg.
- Musical Performance - Symphony, Band (Trumpet).

Resume upon request. Relocation acceptable.
Contact: Robert Fellers, 2616 S. Main, Findlay, Ohio 45840.
Phone: (419) 422-1366 Home. (419) 423-3333 Business.

SYN-AUD-CON SPONSORS

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

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

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

United Recording Electronics Industries
HME
Shure Brothers, Inc.
Sunn Musical Equipment Company
Crown International, Inc.
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Rauland-Borg Corporation
Community Light & Sound, Inc.
Industrial Research Products, Inc.
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