SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

P.O. Box 1239 Bedford, IN 47421
Ph: 812-275-3853

newsletter

Volume 14, Number 1
Fall 1986
©Don & Carolyn Davis

A FIRST

IN AUDIO

IS HARD

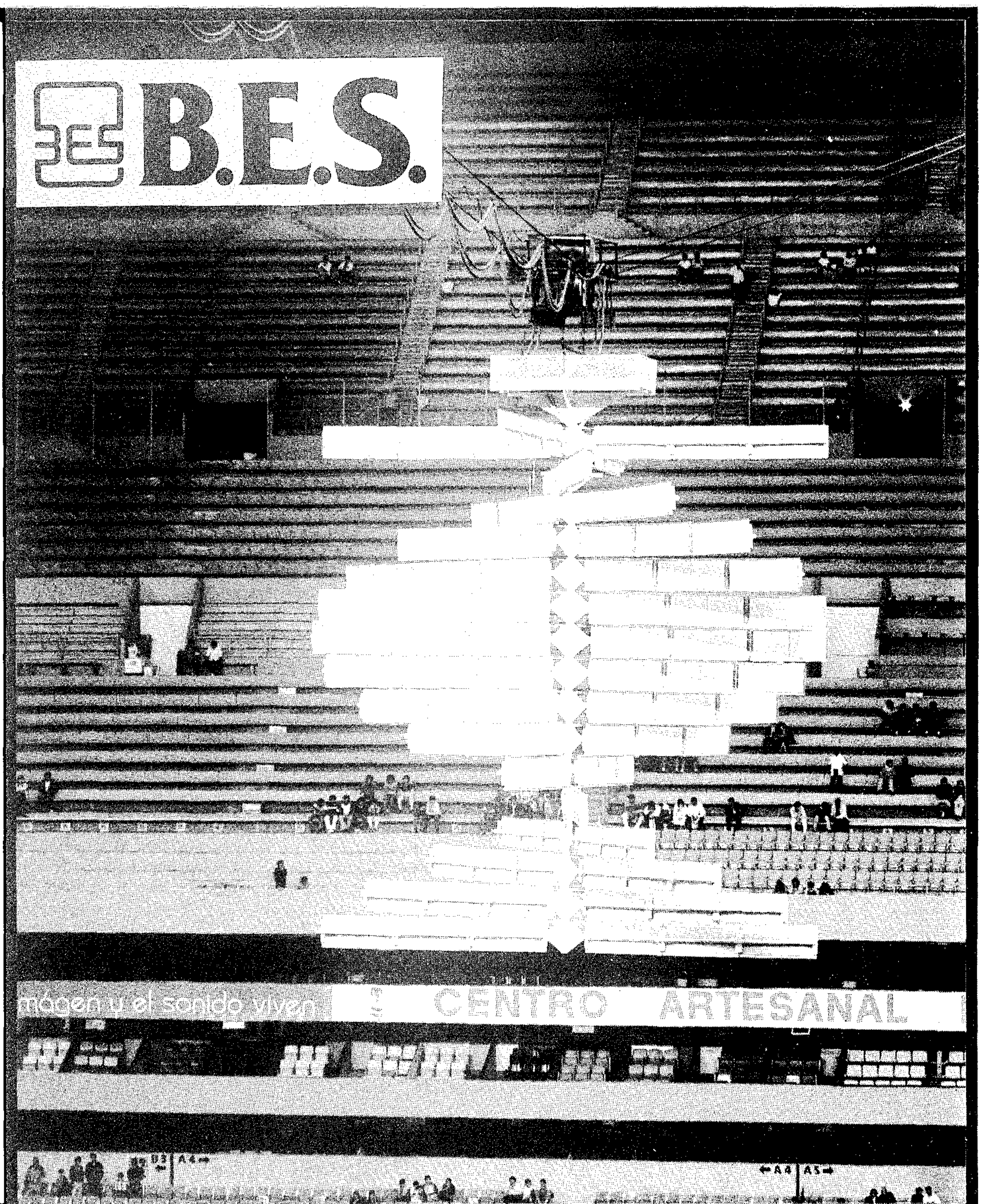
TO ACHEIVE.

HERE IS A

UNIQUE

SUCCESSFUL

"FIRST".



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SIZE OF NEWSLETTER

This Newsletter is smaller than usual because of the length and importance of our Tech Topics.

B.E.S. - SYN-AUD-CON SPONSOR

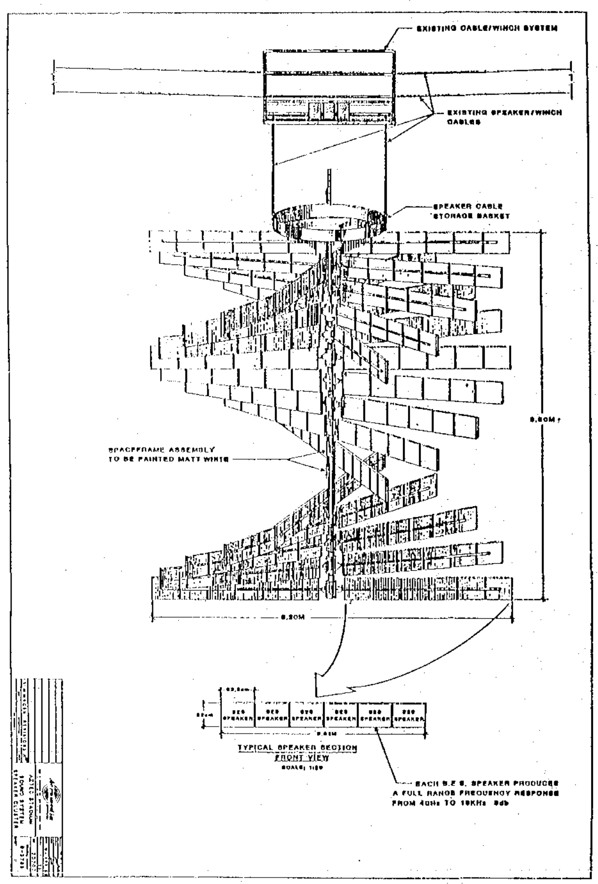
The B.E.S. loudspeaker was designed by Jose Bertagni when he was a professor of physics at the University of Buenos Aires in Argentina. Professor Bertagni's early models appearing in the US market under a variety of brand names including Kenwood and Fisher.

Basically a driven resonant panel system with bipolar high frequency response, nearly omni-frequency response at 1800Hz and reasonable frequency range for musical purposes, the B.E.S. systems - offer some unique advantages to sound contractors:

1. Their flat configuration makes them ideal for ceiling mounting. Their C-70 series is made to match standard ceiling tile. This flat format is a perfect choice for under balcony ceiling mounting in signal delayed systems. Currently the C-70 has found wide acceptance for ceiling distributed systems both for aesthetics and quality reasons

2. They are extremely waterproof. We have specified them in saunas

3. Their key personnel is alert and receptive to new and daring uses of their product in either regular or modified form. (See the front cover of the Newsletter.)



Acromedia of Culver City, Ca, working with engineers at B.E.S. designed a central cluster for the opening ceremonies of the World Soccer Cup tournament which was held in June in Mexico City. The games opened in the gigantic 128,000 seat Aztec Stadium with an elaborate music and folk dancing spectacle with all 24 competing countries participating, which meant that music quality was as important as loudness and intelligibility. The system was designed to blend aesthetically with the decor of a very modern all concrete stadium. The design is dramatic in both appearance and use of the B.E.S. panels.

We had seen the drawings of the impressive array at the NSCA show in Las Vegas in late April. After the games we talked to the engineers at Acromedia to see if the system had performed as they expected: 100 dB anywhere in the vast stadium up to 400 feet away from the cluster. Bob Patrick said that they were very pleased and that the only change they had to make in the original design was to remove a few panels because of a weight problem.

Rolly Brook of BBN in Los Angeles has used a B.E.S. loudspeaker for his auditorium reverberation measurements. Recall that higher Q devices give misleading early decay times. Rolly likes their small size, light weight, and ability to excite the entire hall over the frequency range from 125 to 5000Hz (Their frequency range is greater than this but S/N constraints at the low end and practicality at the high end recommend the range mentioned.)

If the high fidelity rhetoric surrounding the promotion of these devices has put you off, now is the time for a second look. (When we told David Klepper we were using a B.E.S. speaker as the low Q speaker for our Intelligibility Workshop, he said, "Why, they are a high fidelity manufacturer?") We heard one consultant at the Workshop say that he was impressed with the quality of the speaker in the Paramount where reverberation and reflections were not a problem to cope with. We are hearing from many satisfied contractors as to their applicability, especially in distributed systems, reliability, and capability. We asked B.E.S. for a list of their reps as that tells how serious a manufacturer is about the professional marketplace. They have a very fine rep organization. We are pleased to have B.E.S. as our latest sponsor.

Take a minute to read the letter from K. L. Ching written after the Intelligibility Workshop. We were impressed. K. L. Ching and Alex Bertagni came to observe and learn. We like working with sponsors that are open and receptive.

Bertagni Electroacoustic Systems Inc is located at 12753 Moore St., Cerritos, CA 90701. Telephones: (213) 926-0201 (800) 592-4644.

SYRACUSE - AUGUST 18-19, 1986

SYN-AUD-CON MOVES

AFTER DECEMBER 20, 1986:

PO BOX 1239
BEDFORD, INDIANA 47421
PH - 812-275-3853

We love travel and we love the West, but to keep our home and office in California means that we have to travel and be away from home 8-9 months out of the year with extended trips of 4 months at a time, sometimes more.

By basing in the middle west we can still spend the winter in the West. During the rest of the year we can plan trips from our Indiana base of a month at a time.

We're both Hoosiers, both country lovers, so we think we will enjoy our new base on our old family farm which has been in the family since the 1850s.

We hope we can put together an office staff in Indiana that is as tender loving, kind and efficient as our staff at Rancho Carrillo. We'll miss Pat, Debbie and Kitty -- and our magnificent mountains.

NEC - AUDIO LEVELS

Jose Arizpa of the MusicStand in Tucson called our attention to the fine print note on page 70-577 of the 1984 edition of the National Electrical Code. The particular section he questioned was 640-5.

640-5. **Conductors.** Amplifier output circuits carrying audio-program signals of 70 volts or less and whose open-circuit voltage will not exceed 100 volts shall be permitted to employ Class 2 or Class 3 wiring as covered in Article 725.

(FPN): The above is based on amplifiers whose open-circuit voltage will not exceed 100 volts when driven with a signal at any frequency from 60 to 100 hertz sufficient to produce rated output (70.7 volts) into its rated load. This also accepts the known fact that the average program material is 12 dB below the amplifier rating - thus the average rms voltage for the open-circuit 70-volt output would be only 25 volts.

The key to reading this section is to recognize that the 100 volt open circuit reading (the highest possible is the reference being used rather than the nominal 70 volts the system is operated at when a load is present. This is an assumption that at least 3dB of regulation is present

$$20 \log \frac{100V}{70V} = 3.1 \text{ dB}$$

$$100 \text{ volts} \quad 10^{\frac{-12dB}{20}} = 25.1 \text{ volts}$$

This fulfills the last statement in this section

*"Thus the average (typical would be a better word) rms voltage for an **open circuit**.70 volt output would be only 25 volts."*

Confusion can result if,instead of 100 volts, the reader were to assume the reference were 70 volts.

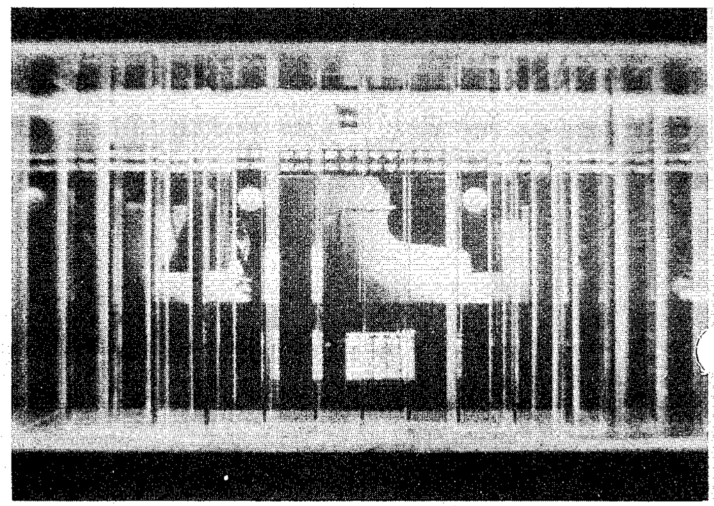
DIFFUSORS AND LIGHT

Peter D'Antonio has built QRDs out of glass where the control room or studio could benefit from having a diffusor and a window in the same location.

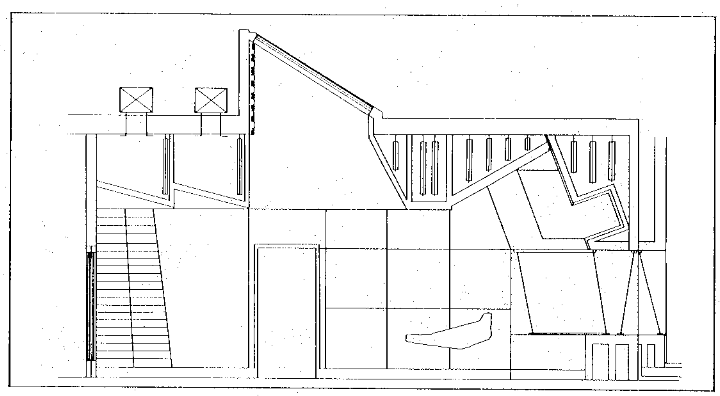
With so many users of recording facilities as visually oriented as they are audio conscious, glass diffusors open an entirely new set of decorative possibilities without the penalty of acoustic compromise.

Russell Berger used the first diffusors that we know of for a window in a studio for Blue Jay in MA. The effect was stunning.

Andy Munro from England is recognizing the value of light in his studio/control room designs. In a recent issue of **Studio Sound** he showed the design of a "lantern" in the ceiling to bring in natural light into the control room. Diffusors are expensive in Europe now. When they are affordable we can expect to see them used very creatively by studio designers.



QRD made of glass for Blue Jay Studio. Designed by Russel Berger



Control room design by Andy Munro. Note the "lantern" in the ceiling to bring in natural light.

MANIFOLDING - A NEW TECHNOLOGY

When was the last time you heard of anything new in low frequency loudspeakers? At least 40 years is a reasonable answer. The horn loaded woofer, both folded and straight, the vented box, infinite baffle, and a host of other solutions for coupling low frequency energy to the air were well developed by 1946.

Now Electro Voice has introduced a genuine new and better way to couple woofers to the air. It's call Manifold Technology[tm]. EV states "This manifolding technique results in increased acoustic loading, yielding **increased low frequency efficiency** and reduced distortion over conventional direct radiating designs in a remarkably small enclosure".

Their specifications indicate that such is the case. Remembering that sensitivity **is not efficiency but that with a knowledge of Q we can obtain efficiency by using**

Loudspeaker Efficiency (%Effic.)

$$\%\text{EFFIC.} = 10\ \text{EXP}\left[\frac{\left(\begin{array}{l}L_{SENST}\text{ at}\\ \text{Ref. D and}\\ \text{Ref. Pwr}\end{array}\right) - \left(10\ \text{Log}\ Q\right) + \left(20\ \text{Log}\left(\frac{\text{Ref. D}}{0.928^{*}\text{ft.}}\right)\right) - \left(10\ \text{Log}\left(\frac{\text{Ref. Pwr.}}{10^{-12}\ \text{w}}\right)\right)}{10}\right] \times 100$$

*Use 0.283 m for S.I.

At 60Hz EV quotes a Q = 2.14 and a sensitivity of 102 dB at 1m from 1w.

Since they quote 1600 watts as a long term power handling capacity this unit can generate at 60 Hz:

1600 watts x 0.0925 = 148 acoustic watts

To verify this figure we can use the equation

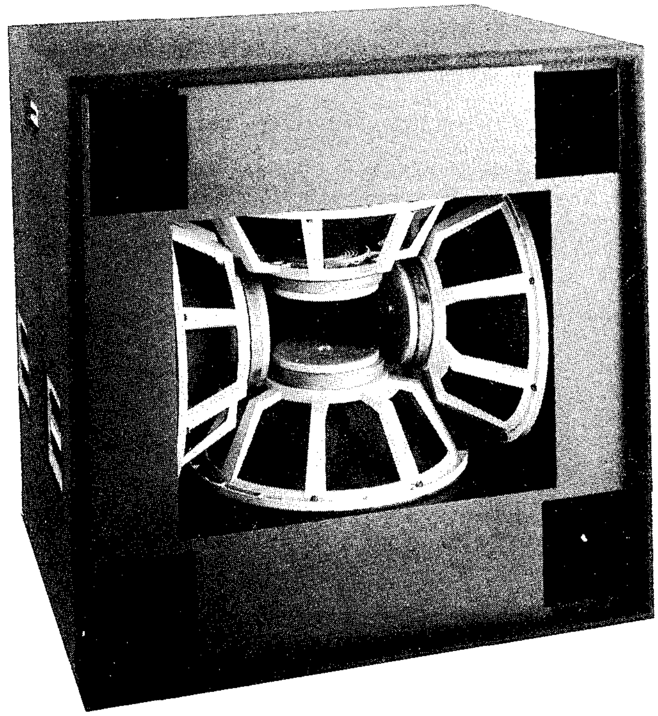
$$W_a = \left(10^{-12}\,w\right)\left(10^\wedge\left[\frac{L_p - \left(10\,\text{LOG}\left(\frac{Q}{4\pi(D_x)^2}\right)\right)}{10}\right]\right)\left[W_e\right]$$

$$W_a = \left(10^{-12}\,w\right)\left(10^\wedge\left[\frac{102 - \left(10\,\text{LOG}\left(\frac{2.14}{4\pi(1)^2}\right)\right)}{10}\right]\right)\left[1600\right] = 148\ \text{WATTS}$$

Their short term power handling capacity (10 msecs) is 6400 watts which would translate into

$$W_a = \left(10^{-12}\,w\right)\left(10^\wedge\left[\frac{102 - \left(10\,\text{LOG}\left(\frac{2.14}{4\pi(1)^2}\right)\right)}{10}\right]\right)\left[6400\right]$$

$$= 592\ \text{ACOUSTIC WATTS}$$



We recently were asked by a grad what devices we would recommend for reproducing gunfire and explosive effects for military training purposes. We replied, the Community M-4 for the upper range and now we can say we know of a real answer to the low frequency range. While heavy caliber rifles and pistols reach as high as 174 dB for 3 msec peak energy at a shooter's ears (centered on about 2 to 4 KHz), the low frequency energy is considerably less. Ten of these units would certainly have me feeling grenade concussion.

MANIFOLDING IS AN OLD ART IN AUTOMOBILES

Our congratulations to EV for actually finding a new way in audio. Manifolding is a high art in automotive technology. Race cars have long employed techniques very similar to what is being employed here. For those interested, read the papers written in the 1960s on the Ford twin overhead cam Indy motor and its unique manifold system.

CLIFF HENRICKSEN

In the last Newsletter we wrote about the new EV DH1A drivers. We credited the design to Cliff Henricksen.

Cliff wrote the following note to correct us about who designed the new EV series of drivers.

> "Please be informed that I did not design the DH1A driver, but rather I directed the engineering team which did. Of course, I had a lot of personal involvement in it. The team included Kent Frye, Fred Digirolamo, Paul Fidlin and Doug Button. I am quite proud of our achievement in this great product."

Our Creator must have loved Cliff Henricksen for when he made him he gave him unique and wonderful qualities, one of which is humility.

SPEED OF SOUND

We are now told by a theoretician, S. K. Wong, that the speed of sound as measured is not so because the theoretician having calculated what he feels is the correct value has decided unilaterally that the measurements are incorrect.

Naturally he receives his main goal: a lot of publicity with his name prominently featured. The error he feels is present is 0.05% lower than the old figure (at 0 degrees C and standard atmospheric pressure he calculates 331.29m/sec whereas the standard has been 331.45m/sec. He now suggests all textbooks, instruments, and formulas be changed to reflect his calculations. What if the publisher left out a letter out of his name and his name is Mr. Wrong.

INTERFACE PREAMP

Allen Burdick, while in our Syracuse class, drew up an application note for a solution to interfacing cassette players, projectors, and similar devices to a commercial sound system.

We are reproducing his handwritten notes here.

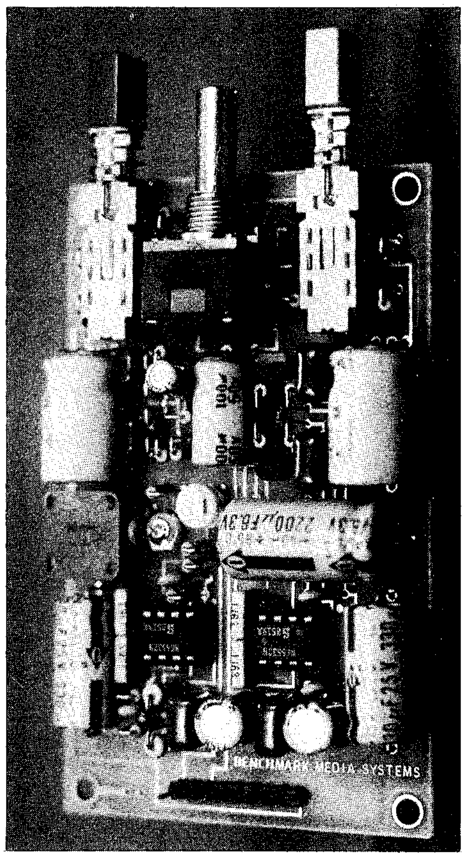
Allen is the owner of Benchmark and the designer of their products. He has in his MIA-4 the lowest noise microphone preamp made by man. While, as he states, the MIA 2 is not as quiet, it is inexpensive and able to serve real needs efficiently.

We have the highest respect possible for Allen's design capabilities. If you have not used his products, you should. It is a pleasure to work with devices that do exactly what it's manufacturer promises.

****

Bal in - Bal out Remote Mic Preamp may be mounted in a double wide electrical box

NF = 11dB



New low noise micriphone preamp

A - OVERALL - ∞ to +64dB

The onboard trim for amplification adjustment (a true gain control) may be removed and a Radio Shack 100K ohm liner taper panel mountpot can be wired in and mounted next to the input XCR allowing overall control of output amplitude. If phantom power is needed the adapter chosen **must** have coupling capacitors to isolate +48V from the input of the MIA-2, otherwise poof!

Allen Burdick of Benchmark addressing the class.

While not the lowest noise device, the cost of the above makes it ideal in many applications not requiring the performance of the MIA-4. This combination will drive 300' of foil shielded cable @ 30pf/ft or 1600' for Mogami 2574 at 6pf/ft.

A line level system may be configured the same way with a DIA-2 set for A = -6dB with the following specifications

Z_{in} = 100 K ohm, Z_{out} = 60 ohm
Max_{in} = +27, Max_{out} = +27
A = to +20dB
THD = 0.002% @ unity & 2KHz

This is ideal for interfacing projectors, cassette players, etc to a system.

POWER REQUIREMENTS

Single + Supply +24 to +36V
@ 50ma Max @ Max output

SYSTEM REQUIREMENTS

Max input - 19dBm (clip)
Max output - +27 dBm (clip)
All @ +36V power supply
THD @ 50 dB = 0.006% 2KHz
Noise figure = 11 dB
Bandwidth 100KHz
Cost approximately $100

NEW MICROPHONE PREAMPLIFIER -- MIA-4

Benchmark has a new Microphone Preamplifier, the MIA-4. This PC board preamplifier is designed for retrofit use and custom applications where a ultra low noise gainblock is required. It has an overall gain range of -2dB through +73dB, a noise figure of 1dB, exceptional RF immunity, balanced output and 200KHz bandwidth at all gain settings. The MIA-4 is designed to operate on dual (+/-) power supplies of 15 to 20V. With 20V supplies and minimum gain the MIA-4 is capable of +32 dBv in and +30dBv out. At 50 dB gain the MIA-4's THD of 0.002% past 20 KHz makes it the only choice for digital recording

Benchmark is a company you should get to know. Benchmark Media Systems, 3817 Brewerton Rd., North Syracuse, New York 13212. Ph 800-262-4675.

SYN-AUD-CON SCHEDULE 1986-87

ORLANDO, FL.
Gateway Hilton
November 18-19, 1986

ANAHEIM, CA.
Holiday Inn
February 3-4, 1987

OAKLAND, CA.
Holiday Inn-Airport
February 11-12, 1987

PHOENIX, AZ.
Granada Camelhead Royale
February 18-19, 1987

STUDIO CITY, CA
Sportsmen's Lodge
March 3-4, 1987

NEW ORLEANS, LA.
Sheraton Inn-Airport
April 2-3, 1987

DALLAS, TX.
The Summit Hotel
April 14-15, 1987

ATLANTA, GA.
The Presidential Hotel
April 22-23, 1987

LOUISVILLE, KY.
Holiday Inn Southwest
MAY 19-20, 1987

DETROIT, MI.
To Be Announced
June 2-3, 1987

BOSTON, MA.
To Be Announced
June 10-11, 1987

SYN-AUD-CON WORKSHOPS

STUDIO DESIGNERS WORKSHOP
Mastersound/Astoria, New York
June 16-18, 1987
(tentative)

FUNDAMENTALS OF LOUDSPEAKER DESIGN
July 1987
(tentative)

INTERNATIONAL CONGRESS ON ACOUSTICS

The 12th International Congress on Acoustics, ICA, 1986 was hosted by the Canadian Acoustical Association in Vancouver, BC. The theme of the meeting was "Acoustics and Theater Planning for the Performing Arts". The dates were August 4-6, 1986 and was easily the most pleasant weather environment on the North American continent during that period. The meeting was graced by some very famous names in concert hall acoustics such as Y. Ando, M. Barron, Jens Blauert, J. S. Bradley, C. Jaffee, L. Kirkegaard, H. Kuttruff, P. A. DeLange, A. Lawrence, A. H. Marshall, J. Meyer, H. Mueller, V. M. A. Peutz, M.R. Schroeder, T. J. Schultz to name but a few.

A 150 page "Proceedings of the Vancouver Symposium on Acoustics and Theater Planning for the Performing Arts" was supplied to each attendee and contained worthwhile new data by qualified workers in this field of endeavor. Our fee for advanced registration was $65 US dollars, which included the above, a continental breakfast the first morning and a marvelous banquet.
And, I almost forgot, a complete listing of attendees, which is very nice to have. We couldn't help asking if the Canadian government subsidized the proceedings to which John Walsh, the coordinating governor of the conference replied, " No, actually we turned a tidy profit". To further increase our pleasure with the conference, a very modest rate was arranged by the conference for attendees at a lovely downtown hotel walking distance to everything, including the Canadian exhibit for Expo.

A significant number of Syn-Aud-Con grads attended (I would say that most everyone who were privileged to have advanced notice of the proceedings -- it was not generally announced. Cliff Sroka told us about it.): Robert Anderson, David Andrews, D. P. Ayyappan, John Bareham, Kenneth Barron, Farrel Becker, Dr. Daniel Commins, Elk Ebert, Don Eger, Grant Klassen, Kurt Graffy, Mary Gruszka, John Laberdie, Tom Mullins, Dan Pacholski, John Prohs, Ron Simonson, Cliff Sroka, Lucille Talayco, Peter Terroux, Xu Yaying.

With a Syn-Aud-Con TEF background the ability to detect which papers had fact and which were chasing fiction was not difficult. We were particularly impressed with Marshall from New Zealand, Mueller, Kuttruff, Meyer and Blauert from Germany, Schroeder from Germany and U.S, Schultz from the U.S. and of course, Peutz.

What's impressive about the list is that they have arrived at conclusions without TEF. What's impressive about Peutz is he arrived at the correct answers years ago and has helped pioneer meaningful TEF advances in applications to room acoustics.

We found this gathering to contain well informed very likable people all working with similar problems and willing to communicate and share. Many competent designers were not present such as Ron McKay of BBN.
The meeting was exceptionally well organized, run with rigor and good humor, and delivered even more than was promised in the promotional literature. John P. Walsh of Artec/ARS Nova Research who conceived of the meeting deserves particular credit for carrying through a spectacularly successful program.

EFFECT OF AN AUDIENCE

Editor's Note. Farrel Becker has made a series of ETC measurements of Wolftrap with and without an audience present. In doing this he has thought clearly about which domain the audience affects. It should be clear to anyone that the audience has **no effect on L_D** but could exert a considerable effect on L_{RE} and L_R. Since equalization can affect only L_D, the question is, "Does our ear-brain system listen to L_D, $L_D + L_R$ or L_T?" For speech the evidence is that we listen **only** to L_D and the first large reflection. We believe that music will prove to be much the same in spite of years of speculation to the contrary. If we adjust L_D with an equalizer the adjustments will, of course, have an influence on L_T. Again, years of experience has shown that where L_D and L_T are very similar such adjustment can be looked at either way (TEF or 1/3-octave). It is our belief that the best way to control audience effects is with controlled polar responses.

Farrel's data is an invaluable "first" awaiting deciphering.

EFFECT OF AN AUDIENCE
By
Farrel Becker

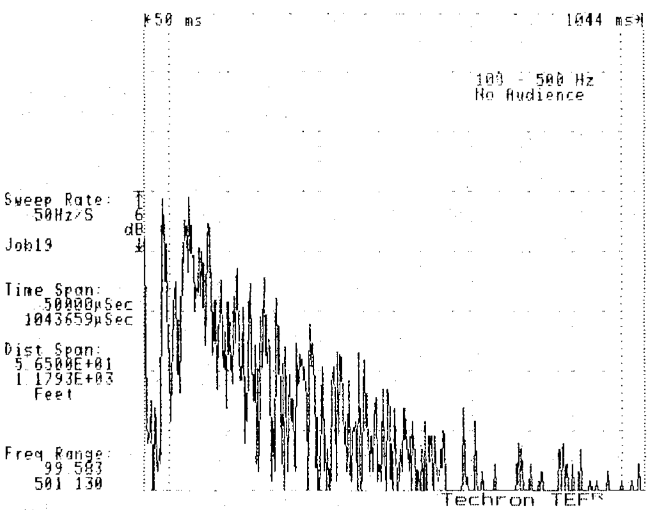
Here are the ETCs with and without an audience. These were the first crude attempts. They were made with Dick Heyser's disk using the vector averaging mode. Each ETC is composed of the vector average of 16 sweeps, each lasting 8 seconds. The total measuring time of each ETC was therefore 2 minutes and 8 seconds plus processing time. It took two 20 minute intermissions to get the ETCs in the five frequency bands.

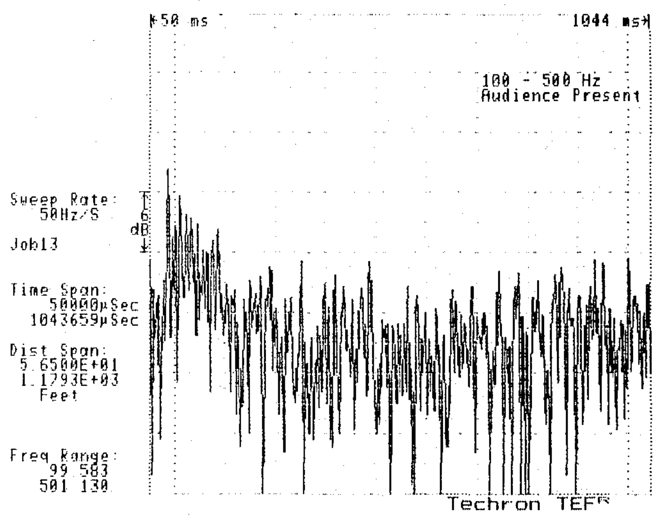
The ETCs with the audience present were made during the intermissions of a Broadway musical. Since the audience was quite noisy, I used an attenuator to adjust the level of the sweep for each frequency range so that the sweep would be just barely audible to me. No one in the audience seemed to be aware of it. You can get some idea of the spectrum of the audience noise by noting how the noise in the ETCs decreases with increasing frequency.

The ETCs without the audience were made the following day. The microphone was in the same location, the hookup and all measurement parameters were the same. The attenuator was set to the same position for each frequency band as it had been with the audience present.

The audience, of course, has no effect on the direct sound. Measuring and equalizing the direct sound, which is what is being done with the SIM® system, is fine if that is what you want to do. But since all that is being equalized is the direct sound, and the audience does not affect it, then why bother to wait until the audience is present? The equalization can be done more quickly (and probably more accurately) before the audience arrives with higher level test signals.

The audience seems to have very little effect in the frequency domain but, as can be seen in the ETCs, has a significant effect in the time domain. The change that we hear is in the time domain and ***cannot*** *be equalized. The effect of the audience on the level and slope of the reverberant field is easily seen, especially at the higher frequencies. This is what is happening when we hear a change in the character of the sound in a room when the audience is present.*





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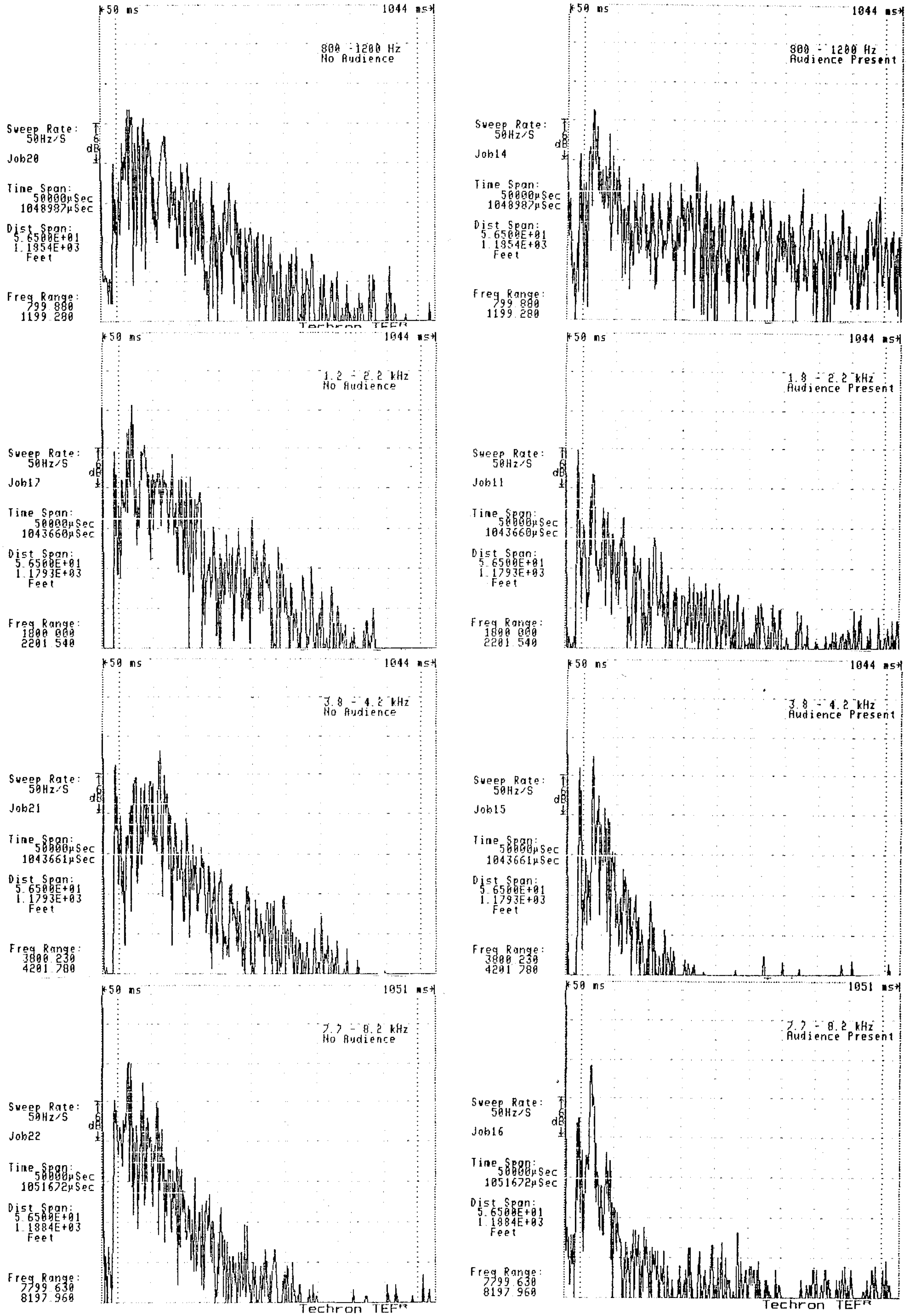

50 ms
1044 ms
800 - 1200 Hz
No Audience
Sweep Rate: 50Hz/S
Job20
Time Span: 50000µSec 1048987µSec
Dist Span: 5.6500E+01 1.1954E+03 Feet
Freq Range: 799.880 1199.280
Techron TEF
50 ms
1044 ms
800 - 1200 Hz
Audience Present
Sweep Rate: 50Hz/S
Job14
Time Span: 50000µSec 1048987µSec
Dist Span: 5.6500E+01 1.1954E+03 Feet
Freq Range: 799.880 1199.280
50 ms
1044 ms
1.2 - 2.2 kHz
No Audience
Sweep Rate: 50Hz/S
Job17
Time Span: 50000µSec 1043660µSec
Dist Span: 5.6500E+01 1.1793E+03 Feet
Freq Range: 1800.000 2201.540
50 ms
1044 ms
1.8 - 2.2 kHz
Audience Present
Sweep Rate: 50Hz/S
Job11
Time Span: 50000µSec 1043660µSec
Dist Span: 5.6500E+01 1.1793E+03 Feet
Freq Range: 1800.000 2201.540
50 ms
1044 ms
3.8 - 4.2 kHz
No Audience
Sweep Rate: 50Hz/S
Job21
Time Span: 50000µSec 1043661µSec
Dist Span: 5.6500E+01 1.1793E+03 Feet
Freq Range: 3800.230 4201.780
50 ms
1044 ms
3.8 - 4.2 kHz
Audience Present
Sweep Rate: 50Hz/S
Job15
Time Span: 50000µSec 1043661µSec
Dist Span: 5.6500E+01 1.1793E+03 Feet
Freq Range: 3800.230 4201.780
50 ms
1051 ms
7.7 - 8.2 kHz
No Audience
Sweep Rate: 50Hz/S
Job22
Time Span: 50000µSec 1051672µSec
Dist Span: 5.6500E+01 1.1884E+03 Feet
Freq Range: 7799.630 8197.960
Techron TEF
50 ms
1051 ms
7.7 - 8.2 kHz
Audience Present
Sweep Rate: 50Hz/S
Job16
Time Span: 50000µSec 1051672µSec
Dist Span: 5.6500E+01 1.1884E+03 Feet
Freq Range: 7799.630 8197.960
Techron TEF

HOW TO MEASURE ALIGNMENT

These instructions are to serve as guidelines only. Alignment is completely new. The instructions are the result of our experience. Don't let the instructions limit you. In fact, let us hear from you if you have suggestions or comments.

Our Sample Case includes long throw and short throw horns.

Alignment Procedures

* Put up a long throw and a short throw horn. Ideally the short throw should be 1/4 the Q of the long throw. Be sure that the mouths of the horns are lined up perfectly - not the acoustic centers

* Listen to each horn with pink noise and with voice one at a time

* Turn on both horns

* Listen to both horns with pink noise and with voice. Make an ETC measurement in the overlap area with a time scale resolution set as high as possible (short full screen time)

* Turn off all but one horn

* Set your microphone off-axis in the pattern of the horn (some devices may require on axis measurement but most do not)

* Make an ETC and an EFC of the horn (print out)

* Turn on 2nd horn which overlaps with 1st horn

* Make an ETC and EFC of the two horns (print out) See Figure 1

* Calculate the out-of-alignment amount from the ETC

* Calculate the out-of-alignment amount in time and dial in the appropriate amount of time on the precision alignment device

* Make an ETC and EFC to see how the curve differs from the original single horn (See Figures 2-4)

* Make an EFC and adjust the signal alignment device until the EFC of the two horns look as close to the original horn (print out)

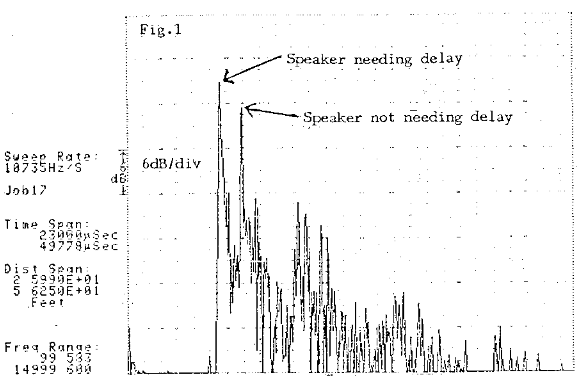
* Make a $\%AL_{cons}$ measurement with TEF and with RASTI if desired

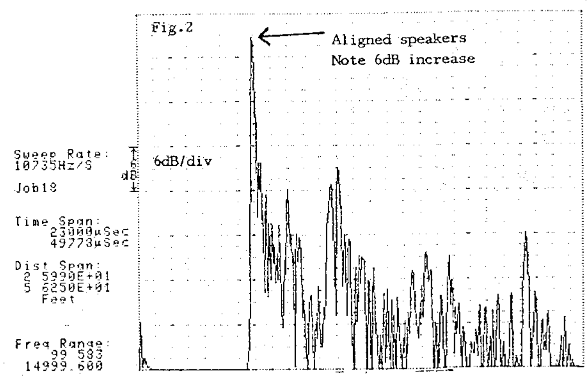
* Listen with speech and pink noise to horns in and out of alignment. Especially listen in the overlap area. For any audible changes in polar response manifested as increased reverberation

We would like to hear from those of you who are aligning horns and speaker systems. Send us your measurements to share in the Newsletter along with your subjective judgments. We can all learn very quickly if we share. All measurements should include before and afters to be considered for publication.

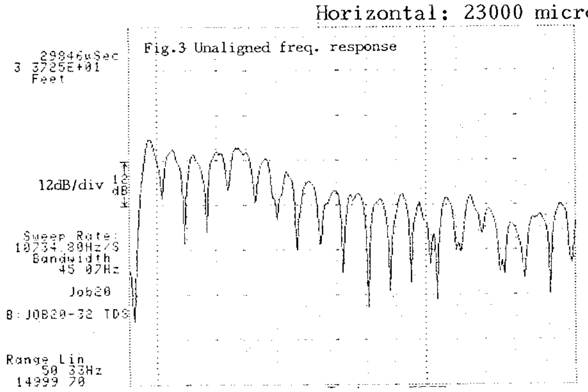
We feel alignment is more important than equalization. It took almost 10 years for equalization to be commonly used, even by the most alert users. There is no reason for alignment to take that long if we all share.

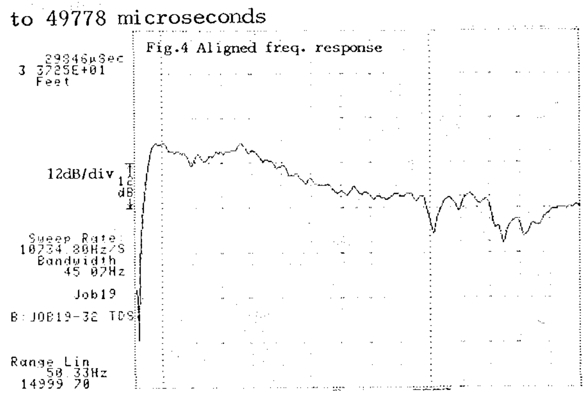
UREI announced their precision alignment device in this Newsletter. We are sure it will be a good unit. Audio Devices in Eugene, Oregon should be in full production with his prototype models.





Horizontal: 23000 microseconds to 49778 microseconds





Horizontal: 50.33Hz to 14999.70Hz

JBL/UREI USEC SIGNAL DELAY

UREI has just made a firm announcement of January 1987 availability of their model 7922 audio delay.

It features:

1. 0 to 327 msecs in 10 usec steps
2. Two independent outputs
3. 16 bit linear conversion (90 dB dynamic range)
4. Linear phase anti-aliasing filter
5. Digital oversampling filter
6. Simple headroom control/indicator
7. Active balanced inputs
8. Transformer outputs
9. Key lock-out security
10. AB display/control for precise array alignment
11. High quality low noise A-D converter
12. A reference output

Very pleasant news is the price of only $1300 (list) for two outputs. The model 7922 is a precision microsecond delay with two outputs as well as a full millisecond (327 msec) delay.

The UREI 7922 should be an excellent unit. We'll be reporting in much greater detail in the near future on how to most effectively use signal delay. Jim Carey, Farrel Becker, Bill Peterson, Randy Vaughan, and we're sure others know how to properly use these tools. We have also witnessed a few who do not. They can be used without a TEF analyzer but its sure not advised. Those of you wishing to share your alignment measurements in the Newsletter, please be sure to include either a very high resolution ETC of before and after or else a clear cut EFC of before and after.

We sincerely believe that precision signal delay compensation will rival the impact that equalization had when first introduced twenty years ago. The 1/3-octave real time analyzer was used to make what the equalizer was accomplishing visible. The TEF analyzer will be key to the use of precision signal delay.

THE ENHANCED ORCHESTRA

The June issue of RE/P had an article, **Electronics & the Symphony Orchestra.** The article discussed the fact that Philadelphia Orchestra conductor Riccardo Muti is using on stage sound reinforcement for some of the instruments. The speaker systems, direct radiating, were designed and built by very capable designer, Kenton Forsythe, director of engineering for Eastern Acoustic Works.

Muti is quoted as saying that "...we maybe seeing the first light in a dark night--the beginning of the Future of Music".

The article notes that the sound system is currently used for sound effects only but as Ken Berger, EAW president, says, "It is now an accepted part of their instrumentation. The creative possibilities now open to modern composers are very interesting to ponder."

The article sites several firsts for the Philadelphia Orchestra in its long and illustrious history but it missed one of its most significant accomplishments: the enhancement of the Philadelphia Orchestra in the 1931.

We have written on it several times in the Newsletter and most recently in V13N1, **William B. Snow & the Enhanced Orchestra** we wrote of our having Bill Snow to lunch at our house in Tustin when he had retired from Bell Labs.

Quoting from our article:

"Bill Snow's discussion of the 1931 stereophonic broadcast was most memorable. He said that the recording and broadcast is remembered for its stereophonic aspect but what was most significant to him ....was that the orchestra was enhanced by 3 dB. Leopold Stokowski was the conductor. Stokowski wrote in his book entitled ***Music For All of Us*** *(-Simon & Schuster, New York, 1943) `When electrical instruments are relatively perfect, they will free musicians from our present constant pre-occupation with the imperfections and technical difficulties of instruments. We shall be able to give all our feeling and thought to the inner essence of the music, because the instruments will respond with extreme sensitivity to every difference of feeling in the player and the music.'"*

Conductor Muti is carrying on the work began in 1931. But, we cannot overlook the work of Pierre Boulez at IRCAM. Boulez resigned as conductor of the New York Philharmonic in the early 70s to head up IRCAM. It is Boulez that is at the cutting edge of computer generated music, electronic music, and electronic reinforcement of instruments and orchestra.

EQUALIZER BASICS

The Newsletters from time to time include short tutorial data on equalizers, microphone, loudspeaker and other audio devices. This one relates to equalizers and how their operation affects the phase and time response of a **system.** Most misinformation on equalizers that we encounter stem from looking only at what the equalizer does. What we really need to know is what the **system** does when the equalizer is added to it.

A PARAMETRIC EXAMPLE

We are using as our example for this test a parametric equalizer.

Biamp makes a very fine parametric, the EQ140. Figure 1 shows the amplitude and phase response of the EQ 140 in its narrowest band pass mode. Figure 2 is the EQ 140 set to its narrowest band rejection mode. Figure 3 is the resultant of both filters in the circuit at the same time. Finally Figure 4 of this series is the magnitude and phase of a 600 ohm resistor. This data tells us that minimum phase response devices when equalized end up with uniform magnitude and uniform phase response.

WHAT ABOUT TIME RESPONSE?

The next set of figures were measured to show the relative differences in time behavior of these four conditions. Figure 5 is the BPF, Fig 6 is the BRF. Figure 7 is the results of both in the circuit at the same time and Figure 8 is the 600 ohm resistor used as a reference.

Figures 9 through 12 are the FTCs of the same data. Here we can see the relative signal delay from introducing the two filters in the circuit at the same time. Figure 12 is the 600 ohm resistor. Figure 11 is the resultant response. Both are measured under identical conditions. Note how much longer relatively it takes the same amplitude to be reached with the filters in the circuit (relatively because we are smearing both time and frequency to obtain this composite view).

From this data we can conclude that minimum phase filters when properly used with minimum phase systems do not introduce any frequency dependent variations in time but actually correct any that are present as a result of minimum phase magnitude variations.

Finally, we can conclude that few parametrics will equal Biamp's performance at any price, and those that will come close will be much more expensive.

PARAMETERS FOR FIGURES 1 THROUGH 4:

Horizontal: 20.35Hz to 1998.10Hz
Log freq axis (2.7 decades)

FIGURE 1. BPF AMPLITUDE AND PHASE RESPONSE

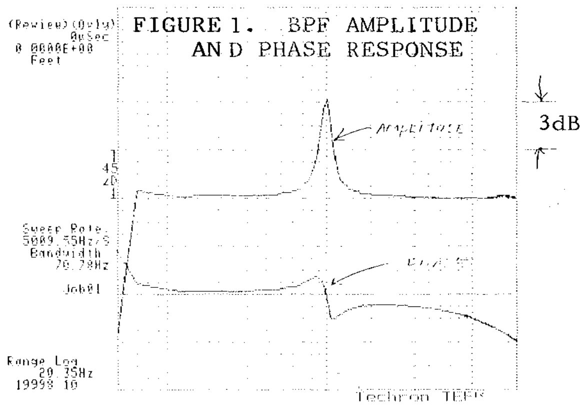


FIGURE 3. RESULTANT RESPONSE OF BOTH FILTERS IN CIRCUIT

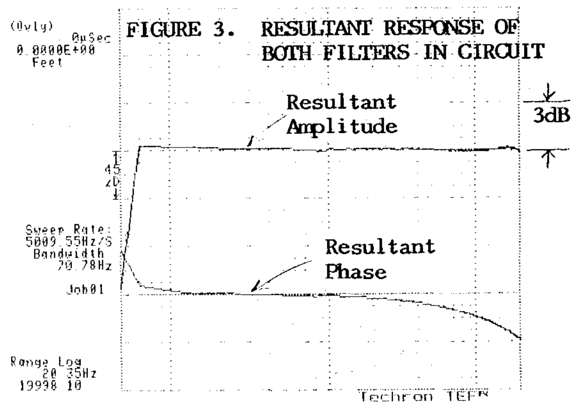


FIGURE 2. BRF AMPLITUDE AND PHASE RESPONSE

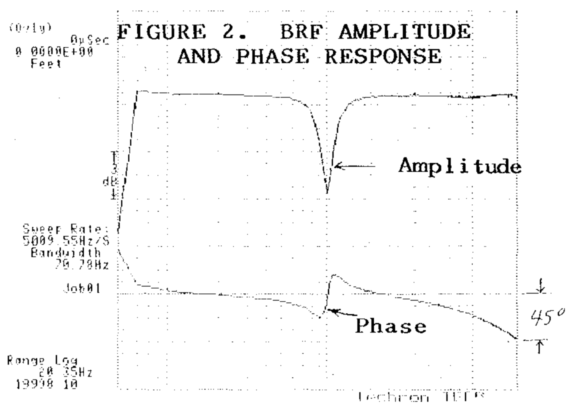
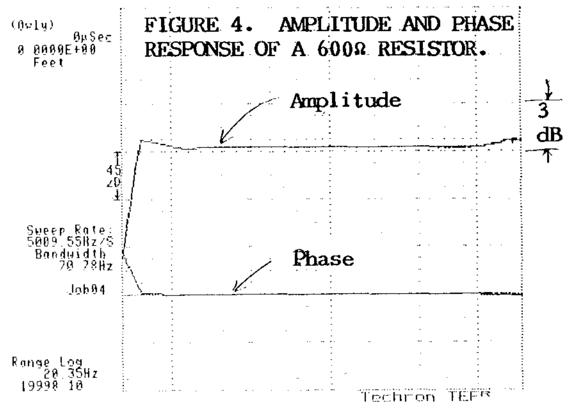


FIGURE 4. AMPLITUDE AND PHASE RESPONSE OF A 600Ω RESISTOR.



PARAMETERS FOR FIGURES 5 through 8:

Horizontal: 20.35Hz to 2000.24Hz

Time of test: 5000 microseconds (front)
to 0 microseconds (back

FIGURE 5. 3-D of BPF.

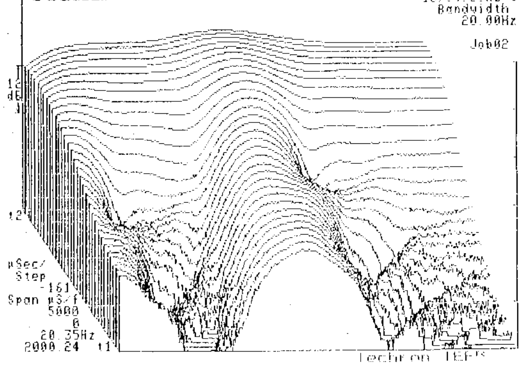


FIGURE 6. 3-D of BPF

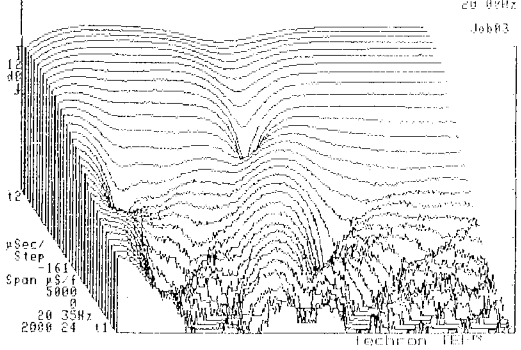


FIGURE 7. 3-D OF RESULTANT RESPONSE OF BOTH FILTERS IN CIRCUIT

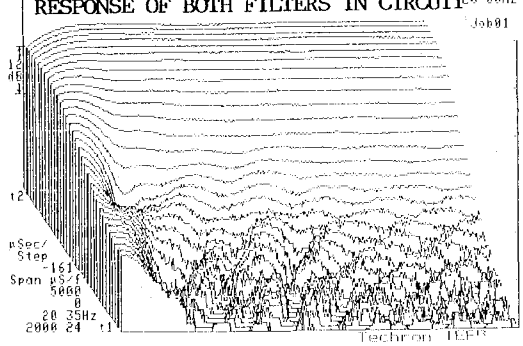
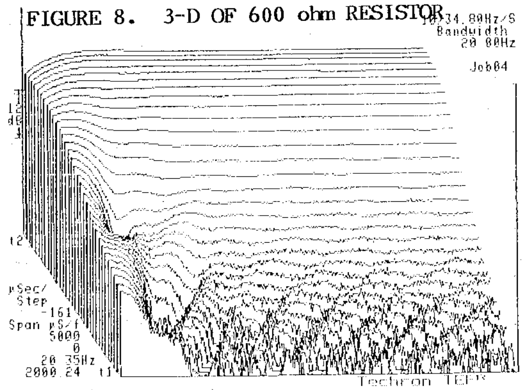


FIGURE 8. 3-D OF 600 ohm RESISTOR



PARAMETERS FOR FIGURES 9 through 12:

Frequency: 20.35Hz to 2000.24Hz

Time: 5000E-6 sec (bottom)
thru: 0E-6 sec (top)

FIGURE 9. FTC OF BPF 3dB/LINE.

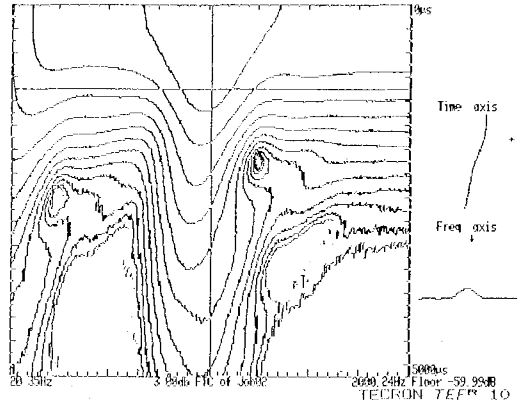


FIGURE 10. FTC OF BRF 3dB/LINE

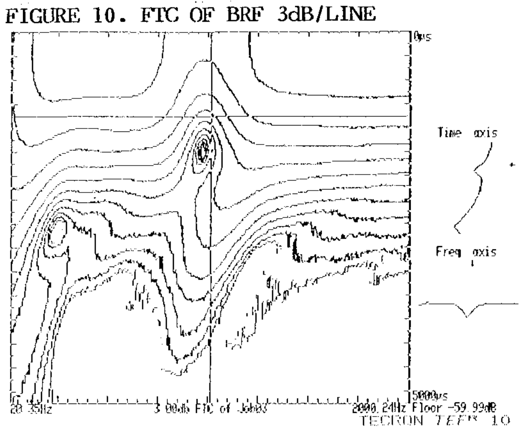


FIGURE 11. FTC OF RESULTANT RESPONSE

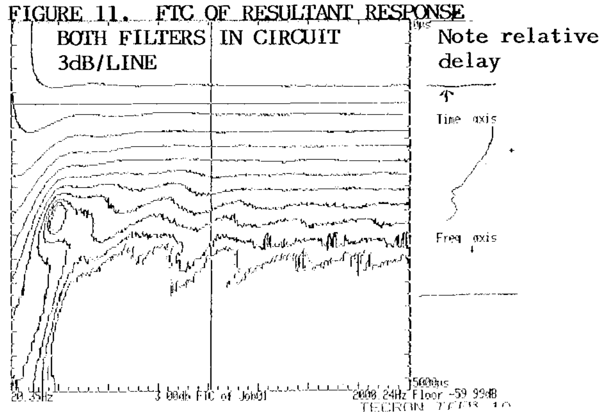
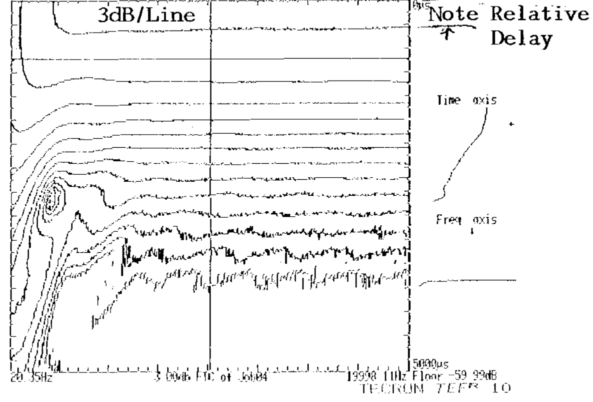


FIGURE 12. FTC OF 600 ohm RESISTOR



STEREO MIXER FROM SHURE

Shure Brothers Inc have designed a professional stereo mixer -- a logical extension of their M267 technology. This new mixer called the FP42 provides two outputs from four input channels, all switchable for mic or line level operation. Each input channel includes a level control, center detented stereo pan-pot, and a pull-pot cueing feature for cueing or checking each input via headphones. It has a concentric clutched master level control. There is also a FP-32 model for ENG work.

SHURE® 85-7-25 9-85 FP42 STEREO AUDIO MIXER

It would be hard to imagine a more useful tool for those jobs where both music and speech must be accommodated. The only suggestion we could offer would be the addition of an A+B and A-B derived third channel output. AC as well as battery operation is standard.

We know from experience how quiet and free from RFI the single channel M267 is. We'd expect the same freedom from such problems in this new unit as well.

K-PADS - A NEEDED ITEM

K-Pads are the small attenuator pads we show in class. Mountings for them come in a variety of forms. The mountings for 1, 2, and 20 K-pads are shown here. Remember that the K-pad itself contains the shunt values for a balanced "H" pad and the mountings contain the series resistances. K-pads are the direct electrical equivalent of the WECO 1c pad and 89 type resistors. K-pads and their mountings can be purchased from Kentrox, 14375 NW Science Park Drive, P O Box 10704, Portland, Oregon 97210-0704. Ph 503-643-1681.

Attenuation in dB	Number of Attenuators	Model Number	Bell CLEI	Catalog Page
211 Block*				
23.0	100	22300103	PD23K007	I-18

Use fixed loss square pads when you can predetermine the required loss and need many identical pads. A back-to-back carrier system is one application.

Kentrox fixed loss square pads have 600 ohm impedance.

Frame mounted 211 blocks with Wire-Wrap or solder terminals are available.* These blocks can be mounted on other frames by using adapters.

TOM HIDLEY ON REFLECTIONS

Interview with Tom Hidley, **Mix Magazine** , August 1986:

"While in retirement, I dreamed of a control room that had no equalization at all. I think I've found the answers, but it was only after a few years of getting away from the business. ***The most important element to consider is the control of first reflections. You have to deal with the ceiling, the walls, and the floor. If you don't have control of the first refections, you will never have naturalness. You may use*** *brute force equalizers for a power balance at a certain point in the room that would be called a flat response - but as you begin to move around, things begin to change. And after a few hours, your ears begin to hurt. Ear fatigue sets in early when there is a high acoustic phase distortion caused by poor first reflections. You get tired, your mind begins to turn off. The power levels have to be kept more restrictive when you have first reflection problems. In these new rooms we're building you can sit and listen with 120 SPL at the mixing desk and know it's loud, but it doesn't kill your ears. In older rooms, 112 SPL is ear shattering, painful."*

ANDY MUNRO

Andy Munro from London has had his TEF analyzer for about a year and a half and he has also been "on his own" for about the same length of time. He was with Turnkey building studios in Europe.
Judging from the number of articles in recent recording studio magazines, Andy is being very successful.

The very beautiful Puk Studios in Denmark is being written about in several of the magazines. More recently **Studio Sound** (July) had a write up on Werner Studios in Denmark. The entire article is interesting but several things particularly caught our interest. One, the effort made to introduce light into the control room (more on this subject in the Newsletter - Light and Diffusion); the second has to do with the interference caused by nearfield monitors on the console:

> "In order to avoid interference with the sound path, the nearfield monitors, a pair of JBL 4312s sits on a black cantilevered shelf and **when required are hydraulically raised** to emerge dramatically from behind the console."

Really clever!

Andy Munro at the TEF Class in Hamburg, Germany.

INFRASOUND & LOW FREQUENCY VIBRATION

In almost every class we have someone ask about the book that discusses low frequency vibration and its effect on the human body. Since we have to look up the reference so many times we decided that we should reprint our original article from V7N1 Newsletter.

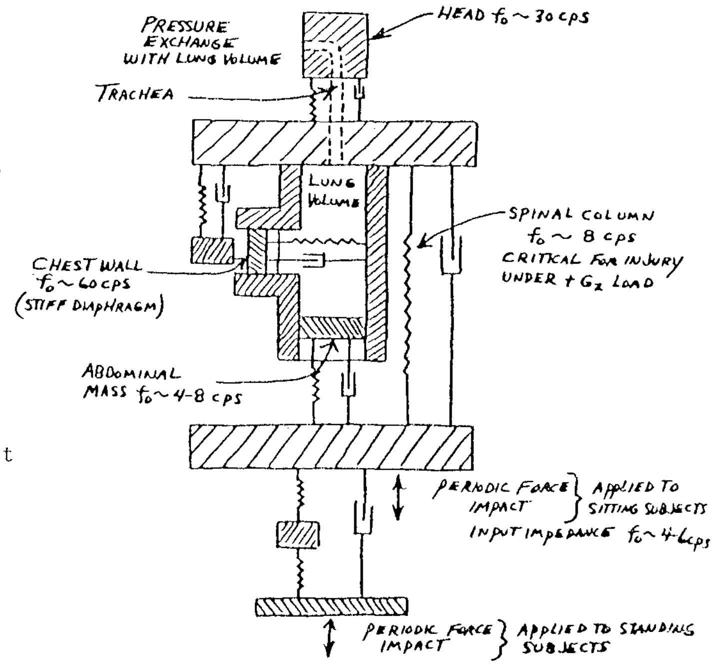
INFRASOUND AND LOW FREQUENCY VIBRATION edited by W. Tempest, Academic Press, 1976.

This 364 page $30 book is exhaustive. (The book was actually ordered by mistake but we found so much interesting material we kept it.) It covers environmental infrasound: its occurrence in measurement, infrasound in transportation, thresholds of hearing and loudness for very low frequencies, physiological and psychological effects of infrasound at moderate intensities, effects of intense infrasound on man, effects of sound on the vestibular system, subjective effects of vibration, the occurrence and its effects on performance of low frequency vibration, vibration and visual acuity, motion sickness and associated phenomena plus an appendix of the international standards in the vibration field along with a useable index.

Replete with excellent illustrations, (see sample) a detailed chart on page 212 of this volume reveals for a standing subject bowel bladder pressure is most disturbed by a frequency of from 10 to 27 Hz. Buttocks and thighs at 2-8 Hz, and chest at 2-12 Hz.

An even more explicit book than the *Shock and Vibration Handbook* by Harris and Crede, we believe it to be a most useful reference for any researcher into sound effects and other effects for theater, discos, etc.

LUMPED PARAMETER MODEL OF HUMAN BODY



CONVERTING EUROPEAN SENSITIVITY RATINGS

Europeans have adopted a sensitivity rating for loudspeakers that specifies the wattage necessary to produce one pascal sound pressure at a distance of one meter. (w/Pa/m). One pascal when expressed as a sound pressure **level** Lp is 94 dB.

$$20 \log \left(\frac{1\text{Pa}}{0.00002\text{Pa}} \right) = 94\text{dB}$$

Conversion # 1

To convert from watts necessary to produce one pascal at one meter to decibels produced by one watt at one meter we use:

$$\text{dB/w/m/} = 94\text{dB} - 10\log\ (\text{w/Pa/m})$$

Conversion # 2

To convert from w/Pa/m to dB/w/4', we use

$$\text{dB/w/4'} = 92.28\ \text{dB} - 10\ \log\ (\text{w/Pa/m})$$

Conversion # 3

To convert from w/Pa/m to dB/0.001w/30' (i.e., the EIA rating) we use:

$$\text{dB/0.001w/30'} = 44.79\ \text{dB} - 10\ \log\ (\text{w/Pa/m})$$

The Inverse Equations

$$\text{w/Pa/m} = \frac{10\text{EXP}(94/10)}{10\text{EXP}((\text{dB/w/m})/10)}$$

$$\text{w/Pa/m} = \frac{10\text{EXP}(92.28/10)}{10\text{EXP}((\text{dB/w/4})/10)}$$

$$\text{w/Pa/m} = \frac{10\text{EXP}(44.79/10)}{10\text{EXP}((\text{dB/0.001w/30'}/10)}$$

What benefit the Europeans feel w/PA/m delivers is not known by us. We still prefer the utility offered by the EIA rating because of the ease with which we can merely add electrical dBm to acoustical L_p.

Once again, sensitivity ratings **are not** efficiency ratings. Relative differences in efficiency can be found by dividing w/Pa/m by the Q at the frequency of interest.

$$\text{Relative efficiency} = \frac{\text{w/Pa/m}}{Q}$$

For an example, if we had a loudspeaker that had a w/Pa/m rating of 0.21 watts and a Q = 10 and another with the same w/Pa/m = 0.21 watts but with a Q = 5, then

$$\frac{0.21/5}{0.21/10} = 2.0$$

We could say that the one with a Q = 5 was twice as efficient as the one with a Q = 10.

We could, of course, do the same thing with the 4'1w ratings.

$$(99\ \text{dB} - 10\ \log\ 65) - (99\ \text{dB} - 10\ \log\ 10) = 3.01\ \text{dB}$$

and

$$10\ \exp\ (3.01/10) = 2$$

Converting your regular specification to w/Pa/m might just be a neat way to confuse competitors that don't know their decibel.

TEF USERS CORNER

We realize that the Newsletter has an abundance of TEF measurements. Many of the measurements are used to put across a very important point - like signal alignment of loudspeakers in an array. The reason for including the TEF measurements is not that we want to sell anyone on the idea that TEF measurements are essential but that alignment is essential, an extremely important concept, one that should be of major interest to consulting engineers and sound contractors.

We are starting a special section of the Newsletter for TEF owners. It will include new programs and how to operate them.

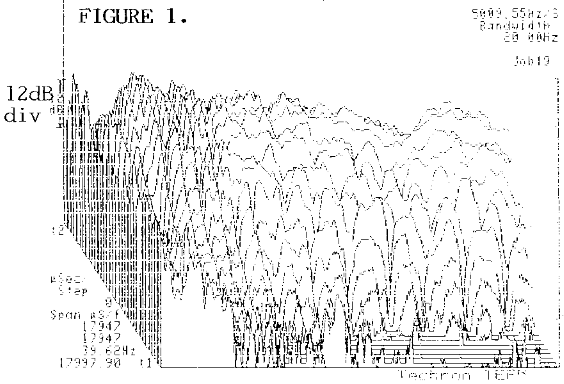
If you don't own a TEF analyzer you can skip the section unless you have some special interest in knowing more about TEF.

You will know that when you see TEF measurements in other parts of the Newsletter that they are there to teach and inform you about something very useful to you. The measurements are worth your time to study.

A NEW WAY TO VIEW COVERAGE ANGLES

One of the ways to post process a 3D polar response is to use the frequency time curve (FTC) program where the vertical axis instead of being time becomes rotational angle.

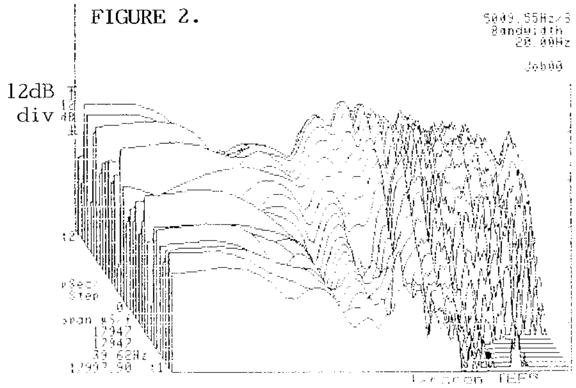
FIGURE 1.



Horizontal: 39.62Hz to 17997.90 Hz
Resolution: 2.5048E+02 Hz

A particularly nice feature of the FTC program is the ability to normalize all data to a chosen reference curve which is made flat (usually the on axis curve). When this is done and the contour intervals are set for 6 dB you see the absolute angular spread at each frequency simultaneously).

FIGURE 2.



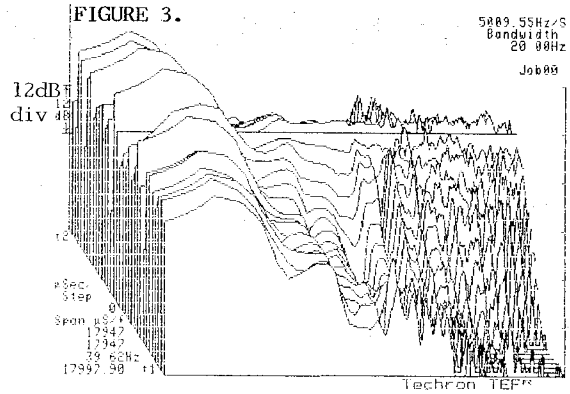
Horizontal: 39.62Hz to 17997.90 Hz
Resolution: 2.5048E+02Hz

It is also possible to post process a 3-D plot either of time or angle into a logarithmic frequency 3-D plot. This can be particularly interesting when viewing coverage angle data.

Figure 1 is of a normal (linear frequency scale) 3-D polar plot. Figure 2 is a post processed log frequency scale 3-D polar plot. Figure 3 is the same measurement but normalized to the on-axis curve of the 3-D polar plot. Figure 4 is the FTC plot with a log scale overlay where each vertical "tic" mark is 10 degrees.

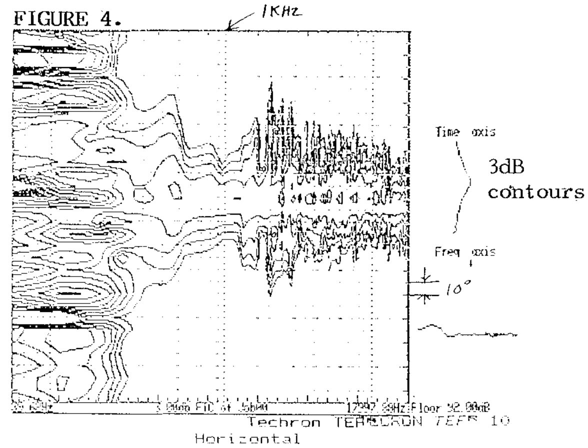
Finding the coverage angle for any frequency merely requires a straight edge that runs parallel to the horizontal base out to the vertical scale "Tic" marks. Simply count the Tic marks and multiply by 10 degrees/Tic mark.

FIGURE 3.



Horizontal: 39.62Hz to 17997.90Hz
Resolution: 2.5048E+02Hz

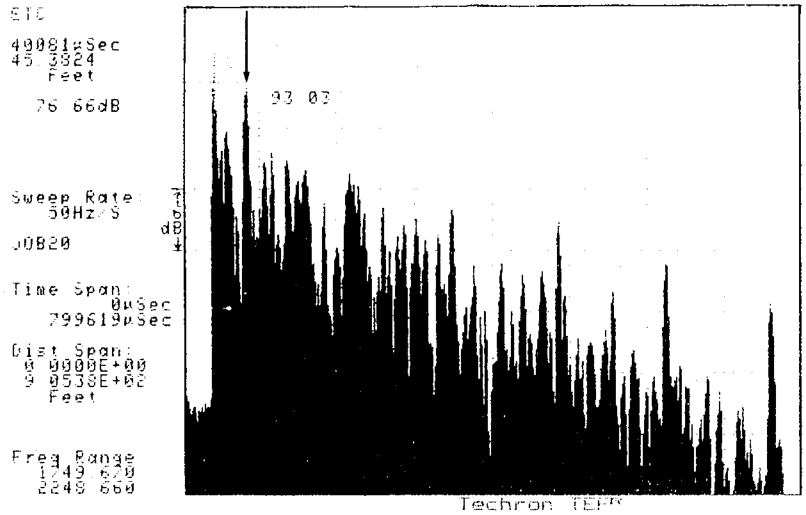
FIGURE 4.

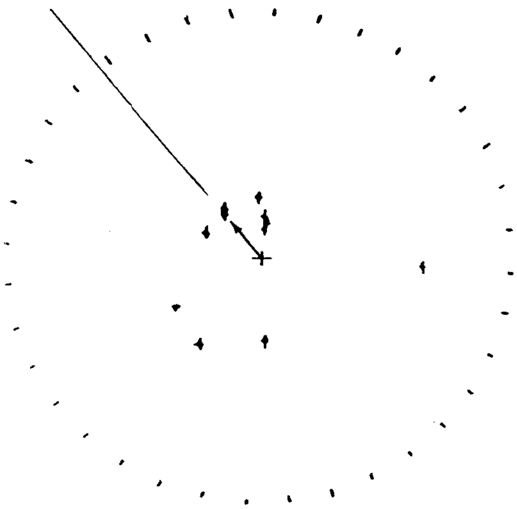


Frequency: 39.62Hz to 17997.88Hz
Resolution: 2.50478E02Hz

THE DIRECTIONAL ETC

Farrel Becker and Peter D'Antonio are rapidly becoming an ad hoc team of signal processing innovators. Their latest collaboration is reproduced here. Such directional displays could have many important uses.





Reflection at 93' was from side wall in Church.

NEW TEF OWNERS

Mike Blackmer
101 Fayerweather
Cambridge, MA 02138
(617) 542-0081

Gis Ingurardsem
Eastside Film & Video
216 E. 45th Street
New York, NY 10017

W. M. Leach, Jr.
Georgia Tech
Electrical Engr. School
888 Hemphill Avenue NW j
Atlanta, GA 30332

Sam Martin
Saturn Corporation
1400 Stevenson Highway
P O Box 7041
Troy, MI 48007
(313) 524-6596

Sheldon Allen
Audio Engineering Co.
10105 W. Hampton Ave.
Milwaukee, WI 53225
(414) 466-4313

Tom Danley
Intersonics
3453 Commercial
Northbrook, IL 60062
(312) 272-1772

W. D. McLaren
Electronic Circuit Specialists
6866 Montgomery St.
San Antonio, TX 78238
(512) 654-7984

NEW LIFE COMMUNICATIONS VIDEO

Craig Hartman of New Life Communications, Willmar, MN, recently sent us a video cassette called **Sound Advice** that they use to inform churches of their services.

It is a professionally made cassette featuring their key personnel, including Ron Huisinga who has attended several Syn-Aud-Con seminars and workshops. The TEF analyzer is prominently used, both before and after installation. They have very convincing testimonials from ministers of churches in which they have worked. It is a very powerful selling tool.

Their brochure contains little touches such as address cards already cut out for use in a Roladex file. They put out their own Newsletter (Sound Advice) and conduct church sound system seminars. The total impact of all this material is to leave the impression of unbounded energy skillfully directed toward the solution of the client's sound system problems.

We were especially pleased to see specific mention of Syn-Aud-Con training as a valued part of their background in the business.

SMILE

A DAY OFF

So you want the day off. Let's take a look at what you are asking for.

There are 365 days per year available for work. There are 52 weeks per year in which you already have two days off per week, leaving 261 days available for work. Since you spend 16 hours each day away from work, you have used up 170 days, leaving only 91 days available. You spend 30 minutes each day on coffee break that accounts for 23 days each year, leaving only 68 days available. With a one hour lunch period each day, you have used up another 48 days leaving only 22 days available for work. You normally spend 2 days per year on sick leave. This leaves you only 20 days available for work. We offer 5 holidays per year, so your available working time is down to 15 days. We generously give you 14 days vacation per year which leaves you only 1 day available for work and I'll be damned if you're going to take that day off!!!

IMPULSE SQUARED TESTING

Three misleading articles on intelligibility appear in the same issue of JASA, September 1986. (The fact that three articles appeared in the same issue of the Journal indicates the timeliness of our Intelligibility Workshop.)

(1) "An Investigation of the Sound Field Above the Audience in Large Lecture Halls with a Scale Model", David W. Kahn and Jiri Tichy. This article contained some truly fundamental measurement errors.

(2) "Predictors of Speech Intelligibility in Rooms", J. S. Bradley.

(3) "Speech Intelligibility Studies in Classrooms" J. S. Bradley. Again poor measurement tools leads to bad guesses.

These articles indicated the rising interest and the dawning realization that the measurement of intelligibility is a present possibility.

Impulse testing can be used within severely restricted parameters. The most commonly used technique is to square the impulse and take its logarithmic amplitude. This superficially resembles an ETC measurements. Unfortunately, it does not show all the energy available at that position in space. Figures 1 and 2 are the impulse squared display and the ETC display. The failure to detect the reflection the cursor is on is evident.

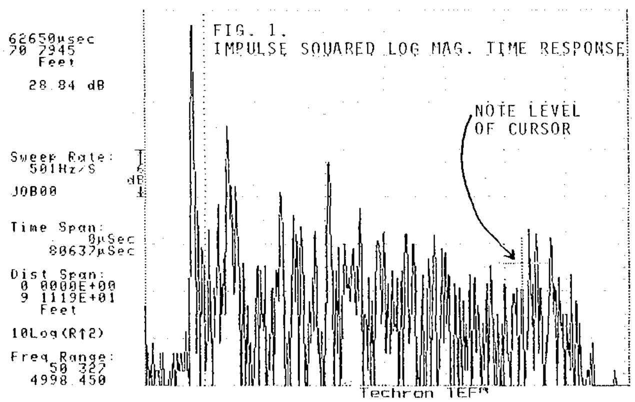


FIG. 1.
IMPULSE SQUARED LOG MAG. TIME RESPONSE

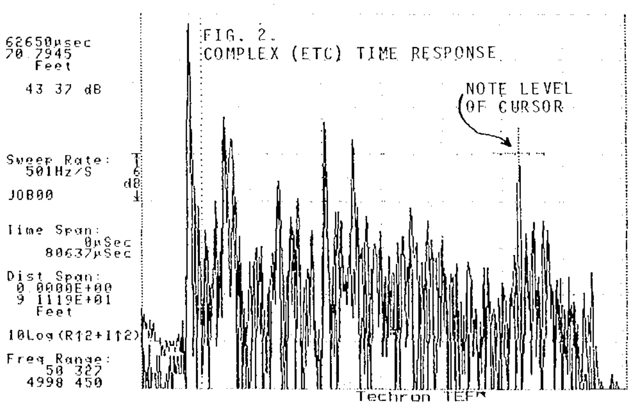


FIG. 2.
COMPLEX (ETC) TIME RESPONSE

To produce a workable impulse model of an ETC (1) the impulse response is squared, (2) **its Hilbert transform is squared and added to it,** (3) the square root of the sum and its logarithmic amplitude is displayed.

IMPULSE TESTS VS TEF TESTING

Current academic teaching utilizes the theoretical concept of the impulse response of a source. The impulse response in the time domain naturally leads to the introduction of the Fourier transform as the mechanism to obtain the magnitude in the frequency domain.

REAL AND IMAGINARY

In class room **theory** this is relatively straight forward. In real life it is a highly impractical way to go about analyzing anything. In **theory**, if you take the impulse response and perform the Hilbert transform you obtain the doublet response. These constitute the so called "real" and "imaginary" parts of the source response in the time domain. If you perform the Fourier transform on each of these you obtain the coincident response (the "real" part) and the quadrature response (the "imaginary" part) in the frequency domain. In the frequency domain it is well accepted among informed engineers that a thorough knowledge of the response of a system requires both parts and that there is absolutely nothing imaginary about the so called imaginary part.

When it comes to the time domain, unfortunately these same well informed people boggle at accepting that both parts are absolutely necessary if we are to see all of the time response of the same signal.

RICHARD HEYSER

Dick Heyser will be presenting a set of papers at the November AES convention in Los Angeles addressing this situation and challenging any and all to try to logically support the validity of using the real part only.

PETER D'ANTONIO

What's in contention here is the common technique of squaring the impulse response and taking its logarithmic amplitude in order to depict the signal distribution over time. Workers like Peter D'Antonio have used TEF analyzers to study this problem. Peter reports to us that he has observed significant signal returns that only appeared in the doublet response and not at all in the impulse response.

BRUEL AND KJAER

Bruel and Kjaer, as usual, does it correctly and measures with their dual channel FFT the impulse response followed by applying the Hilbert transform to obtain the doublet and then squares the quantities, take their square root and displays the logarithmic amplitude as an energy time curve. The theory bothers them too but they don't intend to leave real signal out of their measurements.

Again turning to Bruel and Kjaer literature we find that under the conditions of a source that is amplitude limited (a loudspeaker) and the level of the uncorrelated noise corresponds to the signal level that can be realized using sine excitation then the FFT using impulse response takes 1700 secs (a half hour) for a useful measurement whereas the TEF analyzer takes 0.5 secs for the same accuracy. (See Free Field Techniques by H. Biering and O.Z Pedersen B&K 13.03.81). The answer should be obvious.

DR. PATRONIS

Now a logical question arrives. If the impulse response can be turned into an ETC (if you know how), why should those using impulse response abandon their investment in apparatus and software to obtain a TEF analyzer? As Dr. Patronis puts it, "you can get from town A to town B by walking or you can take a jet. Both ways will get you there but when they are 1000 miles apart, common sense dictates the jet."

CONCLUSION

No serious worker in room acoustics can afford to waste his or her time on impulse testing in spite of the **theoretical** support it received from academia.

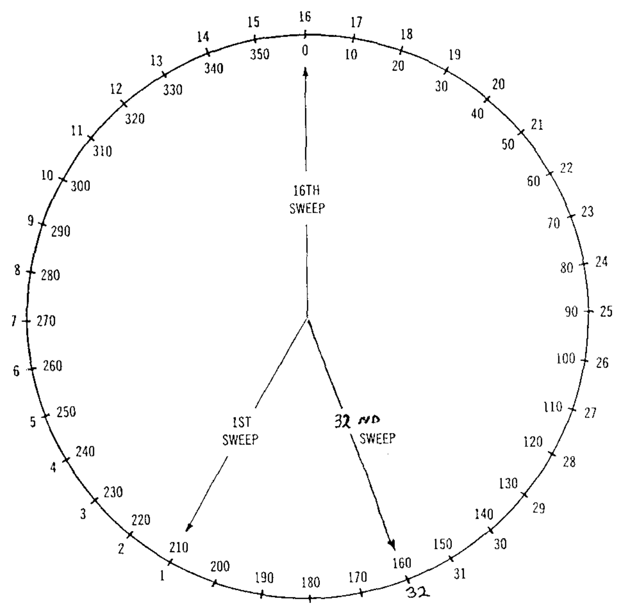
We sincerely hope that Heyser's two AES papers break through this mental roadblock to allow others to benefit from this superior form of analysis and help curtail the flow of faulty data on room acoustics.

TEF POST PROCESSING 3-D POLARS

When polar information on a loudspeaker is made in the 3-D format it becomes a resource for several different post processing procedures on the TEF analyzer. Making such a polar response in each plane of interest is typically handled in the following manner.

DESIRED AND POSSIBLE FREQUENCY RESOLUTION

To make usable 3-D polars requires a test setup that will provide a time window large enough to allow the desired frequency resolution. For example, if you wish 100Hz of F_R then the window $T_R = 1/F_R$ = 10 msecs or a space window of 11.3'.



SWEEP NUMBER (OUTER NUMBERS) RELATIVE TO LOUDSPEAKER ANGLE (INNER NUMBERS)
(MEASURING MICROPHONE ON 0° (SWEEP 16) AXIS)
NOTE 1ST SWEEP ON TEF ANALYZER IS LABELED Ø AND 32ND SWEEP IS LABELED 31.

The point to remember is that when constrained to use a smaller space your ability to step through frequencies on the conventional polar plot program will be constrained to intervals equal to or greater than the frequency resolution. Most of us use a turntable with 10 degree increments which provides a realistic smoothing of minor lobes but still allows the detection of major lobing problems. A few have gone to 5 degree increments and we suspect that will definitely be the point of diminishing return.

For those using 10 degree, Figure 1 illustrates one way of setting up the turntable. The TEF 3-D analysis does 32 sweeps labelled "0" for the first sweep and "31" for the 32nd sweep. When calling for the on-axis curve, it is necessary to remember that the 3-D curve labelled "15" is actually the 16th sweep. This nomenclature can be confusing at first but if you make it your habit to always set up the same way then 3-D curve "15" is automatically the one you'll call up.

If there is a need to have 36 sweeps instead of the 32 normally available, put them on the next job number and after the fourth sweep (the sweep labelled "3"), press "ESC". Go out to A and press B:RTN, which gives you B. You then rename the curves as follows

```
REN JOB 00 - 32.  TDS = JOB 01 - 00.  TDS  RTN
 "   "  00 - 33.   "    "   "  - 01.   "    "
 "   "  00 - 34.   "    "   "  - 02.   "    "
 "   "  00 - 35.   "    "   "  - 03.   "    "
```

Where 00-32, 33, 34, 35 are the new names and 01-00, 01, 02, 03 are the extra four curves.

After re-naming the curves then ERA.d:JOB 01.DSC erases the second job used to temporarily store the four extra sweeps.

A 3-D polar ready to be post processed by either the polar plot program, the FTC program with the plot normalized to the on-axis curve, or called up selectively to plot as EFCs at differing angles. A truly useful tool.

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