SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

newsletter

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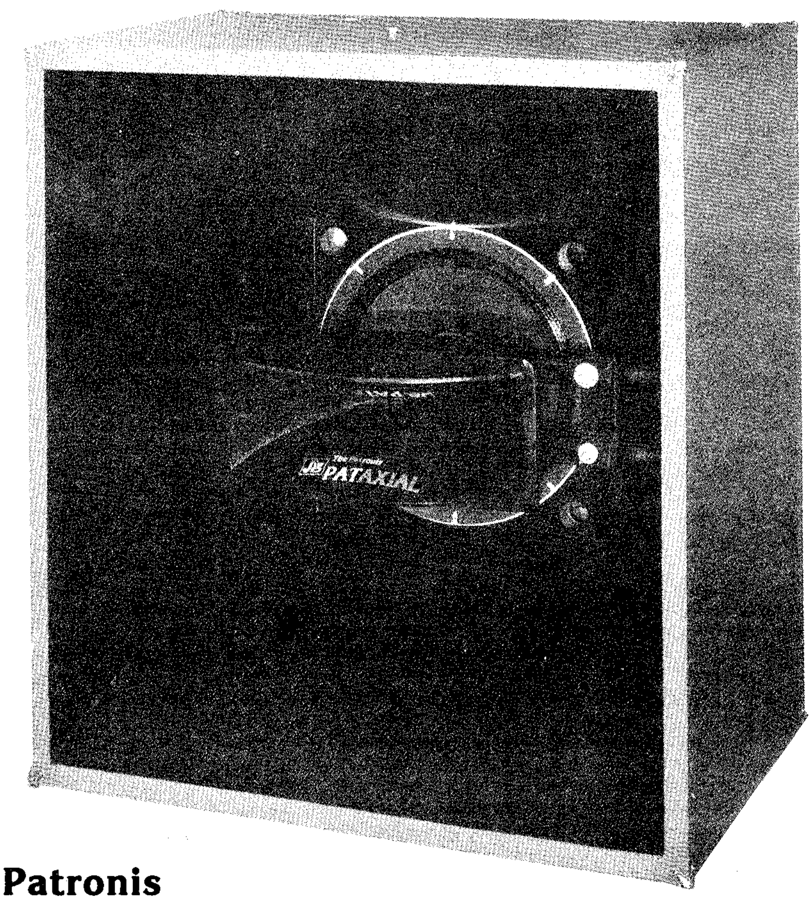
SYNERGETIC
Working together; co-operating, co-operative

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



JBL The Patronis
PATAXIAL

A SUPERIOR LOUDSPEAKER SYSTEM

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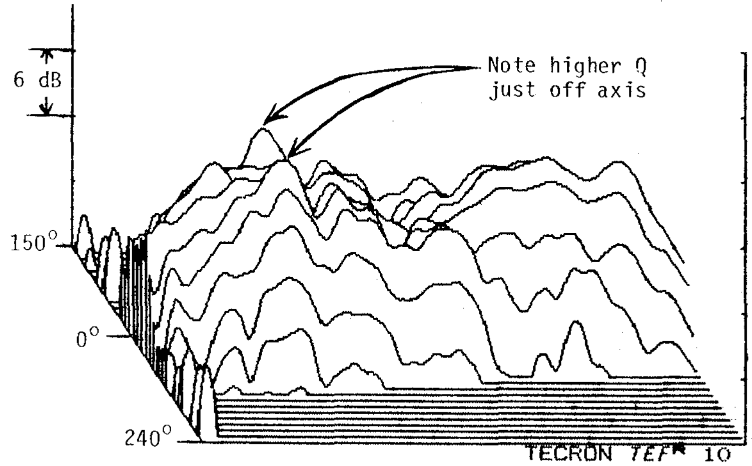
PATAXIAL LOUDSPEAKER

We are in the process of witnessing a remarkable "ad hoc" combination of diverse talents come together in the design of new products. Dr. Eugene Patronis is not only a thorough theorist, a competent consultant, and an extraordinary electronics designer, but he is exhibiting exceptional entrepreneurial facets as well. Manny Mohageri of Emilar, as we have stated in the past, is the toolmaker with the potential to outperform the best of the past.

Harvey Earp of J. W. Davis & Company in Dallas has negotiated with Gene and Manny for the development of a series of truly exciting products.



Vertical: 6dB/div with base of display at 69.3dB
0dB is located at .00002 Pascals

Horizontal: Auto 0.00Hz to 12883.80Hz
scale: 3522.55Hz/inch or 1386.83Hz/cm.

Resolution: 3.3835E+00 Feet & 3.3397E+02Hz

Time of test: 8865 microseconds 1.0017E+01 Feet (front)
to 8865 microseconds 1.0017E+01 Feet (back)
0 microseconds/step or 0 Feet
10°/step 240° through 0° to 150°

Polar plots of Pataxial loudspeaker.

The first of these to be announced is the Pataxial loudspeaker that appears on our cover. Here below are a few of its specifications:

The Pataxial loudspeaker (designed by Dr. Eugene Patronis as a co-axial system) is a loudspeaker truly capable of being described as optimum for both music and speech reinforcement and reproduction.

- Wide frequency range - uniform from 35 Hz to 15,000 Hz.
- Uniform coverage pattern over its useful frequency range.
- Controlled directivity factor, "Q," over the entire speech range with low frequency and high frequency units having the same Q and $C_{\angle}$ at crossover.
- Temporality coincident. The signal alignment built into this loudspeaker system insures the same arrival time at the listener's ears for all frequencies.
- Fully horn loaded for both low and high frequencies and with special signal conditioners that totally remove "horn sound" from this system.

Continued next page.....

- Ruggedly built. The Pataxial system doesn't require additional work on the part of the sound contractor after a "buzz" test with an oscillator -- it's ready to perform as received.
- Real *punch* from a power handling rating of 150 watts broad-band noise 35 Hz to 20 KHz, a controlled Q, and a sensitivity of 96 dB at 1 meter for a 1 watt input (4'/w = 94.28 dB).
- Accurate coverage angles (C_L) of 90 x 40 which may be oriented on either axis as desired.

The 3-D directional response is shown above. It will be quite awhile before you see one as good from any other source.

The first users are ecstatic. Syn-Aud-Con has been the first to tell you about many outstanding improvements and developments in the audio art. We're pleased to be able to again give you the opportunity to be at the beginning on this new generation of products.

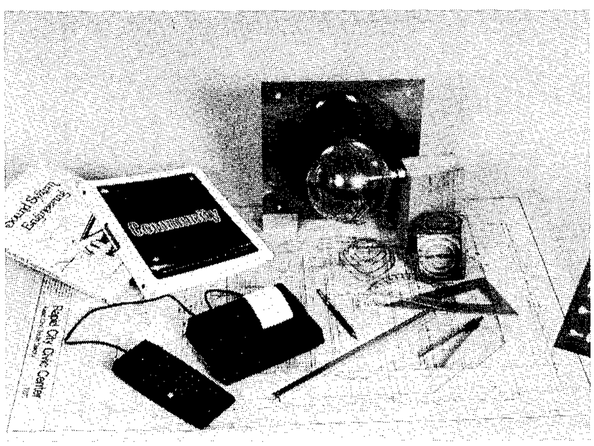
COMMUNITY CLUSTER COMPUTER

1984 will see a number of new techniques come into very widespread application. TEF® analysis is obvious, but the use of the TEF as a 64 K Byte computer for sound system design programs may not be so readily apparent at this time. What makes it apparent to us? The fact that all the necessary subroutines are already in existence and only need to be interwoven into a coordinated program to become the most sophisticated design program available to contractors and consultants.

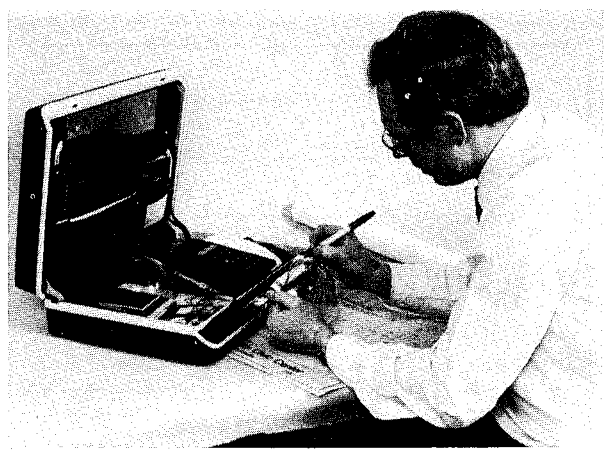
One of the subroutines of such a program is an accurate reliable coverage mapping routine. Community Light and Sound, Inc. will actually be selling their kits for John Prohs' Sphere program by AES time in New York City in October.

Even a casual reader can detect that Syn-Aud-Con's choice is this system. Why do we prefer it to all the other attempts?

1. It is the most accurate system.
2. It is the most versatile, allowing vertical, horizontal and axial rotation to be precisely mapped.
3. It is the only system that allows simple projection onto a screen in order to ascertain if the lineal to angular mapping translates back into the correct room perspective.



Tools for Community Cluster Computer: A Sphere, speaker overlays, and HP-41CV program.



John Prohs mapping onto a sphere.

This last feature is vital. Having observed designers using other programs, it has become immediately apparent that when they make an error in the mapping it is not automatically detectible. When a mapping has been done incorrectly on the Sphere program and is then projected onto a screen, it is painfully apparent in the projection.

If you have not encountered this technique for planning loudspeaker coverage, you are in for the thrill of a lifetime when you finally see it.

Syn-Aud-Con is reproducing the Community order sheet for this Newsletter because we feel that access to this type of advanced design material at such a reasonable price is important to you.

NEUTRIK - Syn-Aud-Con's First European Sponsor

During the summer we had a visit from Bernhard Weingartner of Neutrik AG and Andrew Brakhan of AKG Acoustics, Inc., American distributors for Neutrik AG, at our new offices at Rancho Carrillo. They showed us a very exciting new product and discussed Syn-Aud-Con sponsorship. Syn-Aud-Con is very pleased to welcome Neutrik to its family of sponsors as a company dedicated to professional audio goals.

The product they brought with them is a clear indication of the leadership they intend to demonstrate. It is the Audiograph 3300 measuring *system*. This is, by a wide margin, the most advanced level recorder available today anywhere near the low price of $3400 for a complete system that includes amplitude, phase, group delay, time delay, and RT_{60}. Better yet, it is of modular construction and will be kept up-to-date with other "plug in" modules. The Audiograph 3300 recorder has the best chart handling mechanism it has been our pleasure to work with, and we found the phase module to be easy to use in obtaining accurate phase measurements.

The "group delay" feature is useful in examining electronics and passive networks. It *should not* be used with loudspeakers.

The following measurements were made with the prototype 3332 module while it was loaned to Syn-Aud-Con. We found this module to be exceptionally easy to operate and it makes sound system documentation an easy and accurate task.

Phase delay distortion and envelope delay distortion are often confused. Phase delay is insertion phase shift divided by frequency, ϕ/ω. Envelope delay is the first derivative of phase with respect to frequency, $d\phi/d\omega$ (see Fig. 1). Phase delay distortion and envelope delay distortion occur when phase delay and envelope delay, respectively, vary nonlinearly with frequency. It is possible to have phase delay distortion without envelope delay distortion but envelope delay distortion is always accompanied by phase delay distortion.

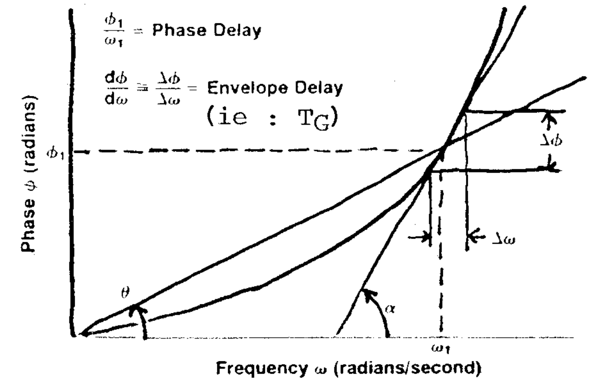
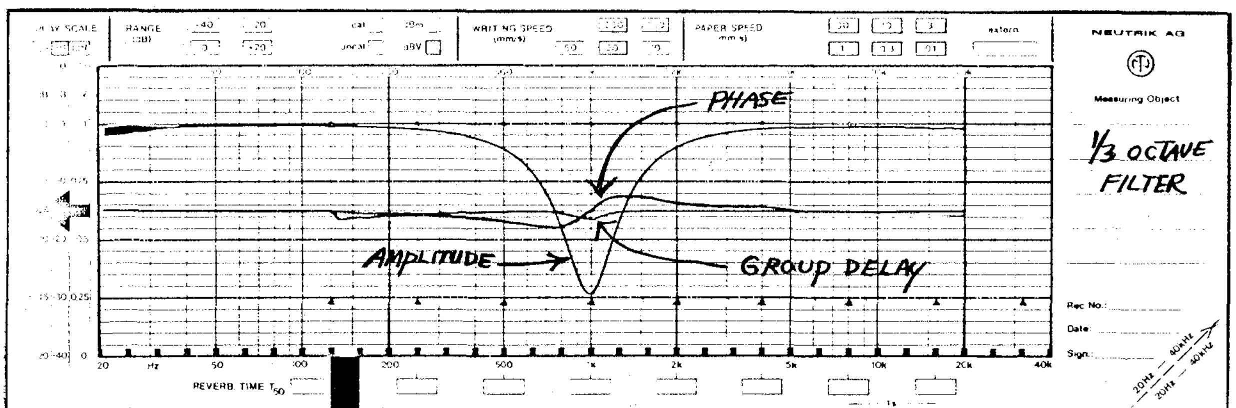
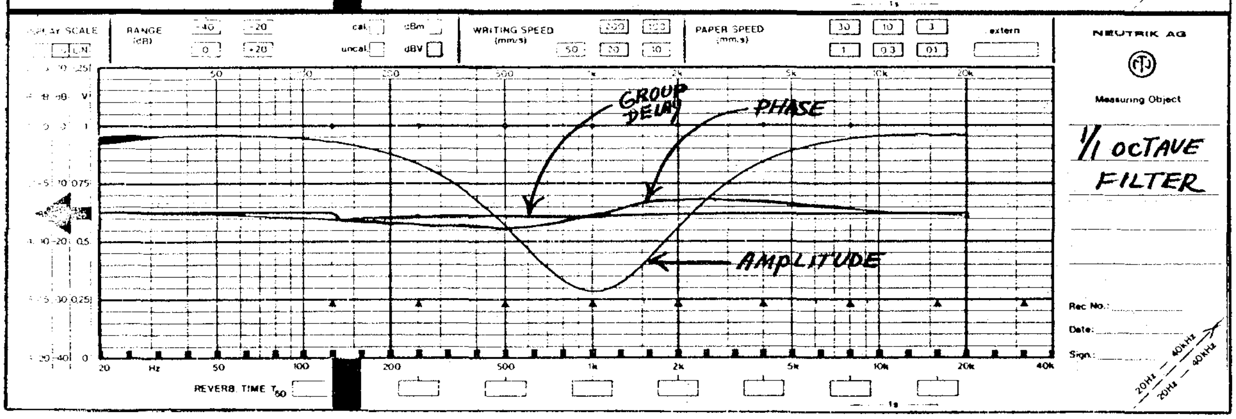


Fig. 1. *Phase delay and envelope delay defined*

From an old Hewlett Packard manual. Drawing has been modified.





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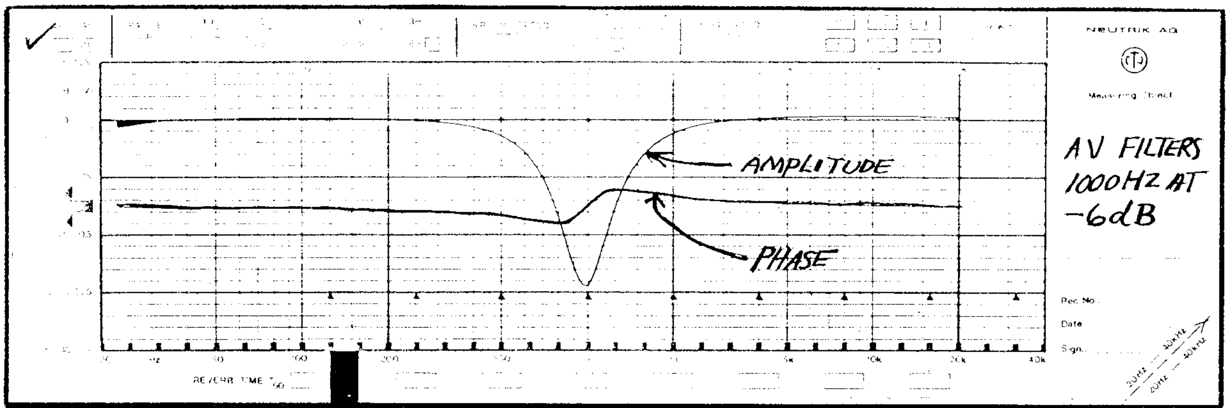

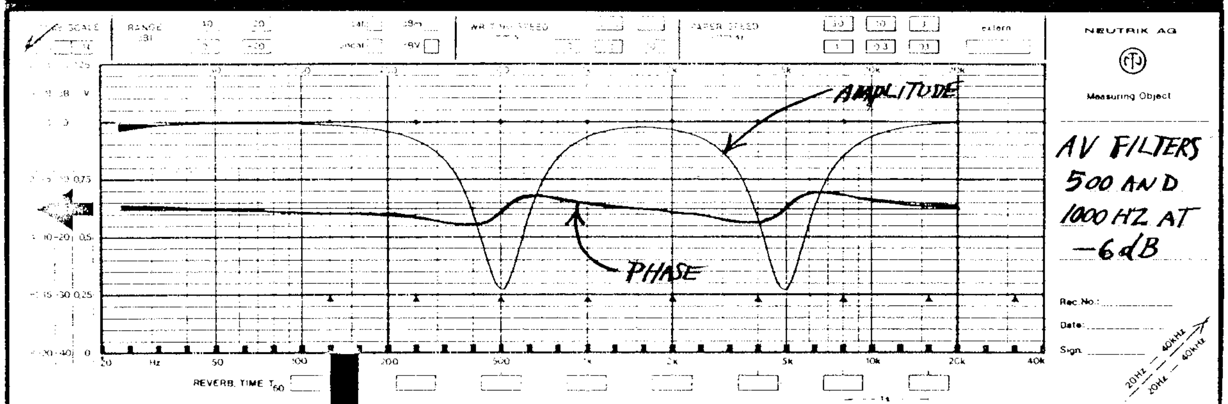

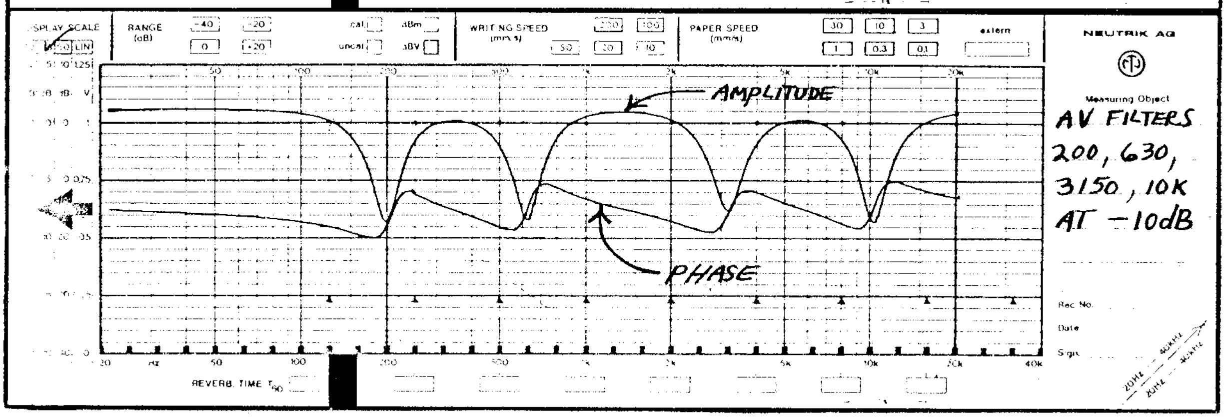

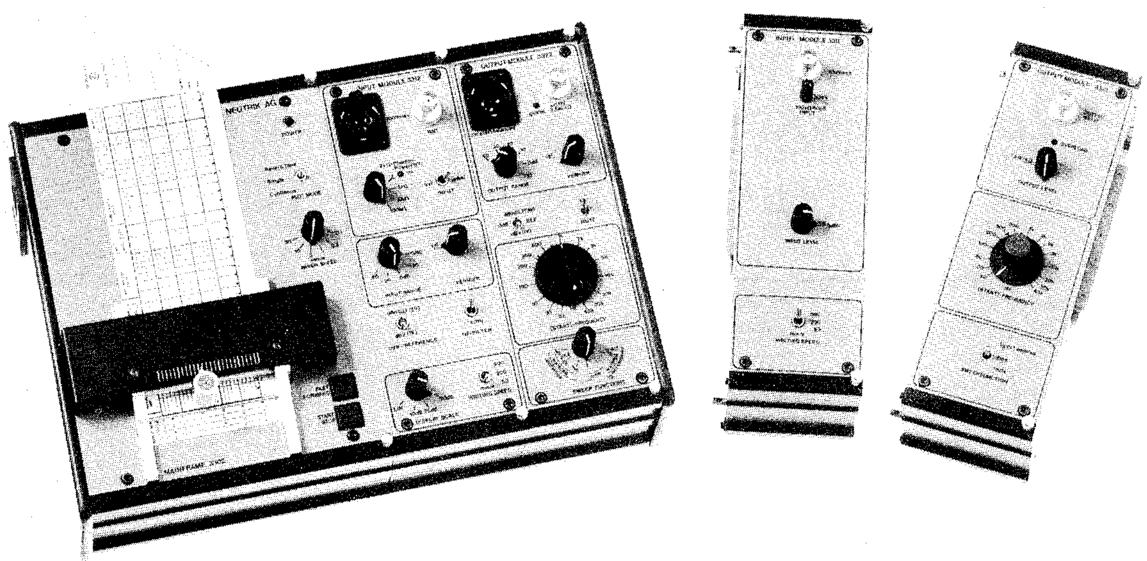

The Audiograph System 3300 shown with the *mainframe 3302* with the 3312 output module already connected to the mainframe module. Shown to one side are the 3311 and 3322 input and output modules for non-professional (i.e., Hi-Fi systems) testing. The phase and group delay measuring module is the 3332.

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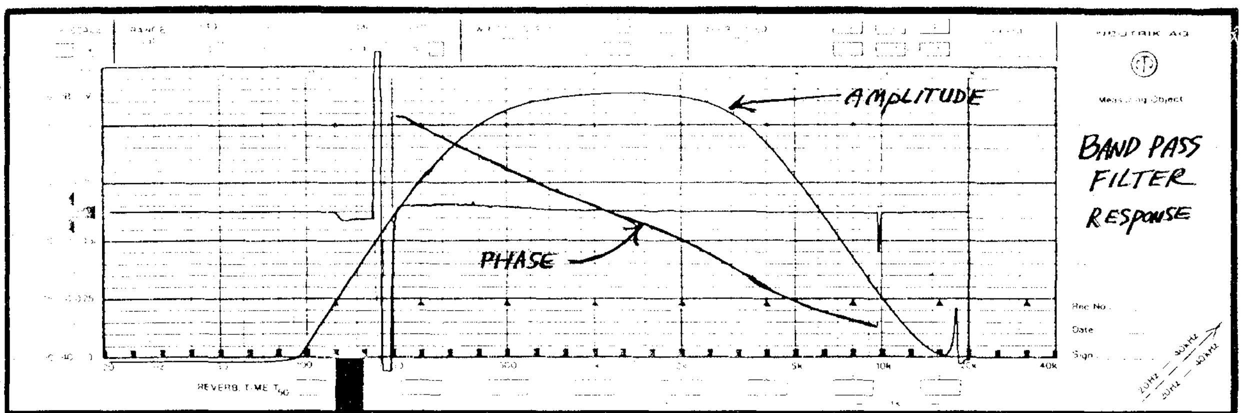
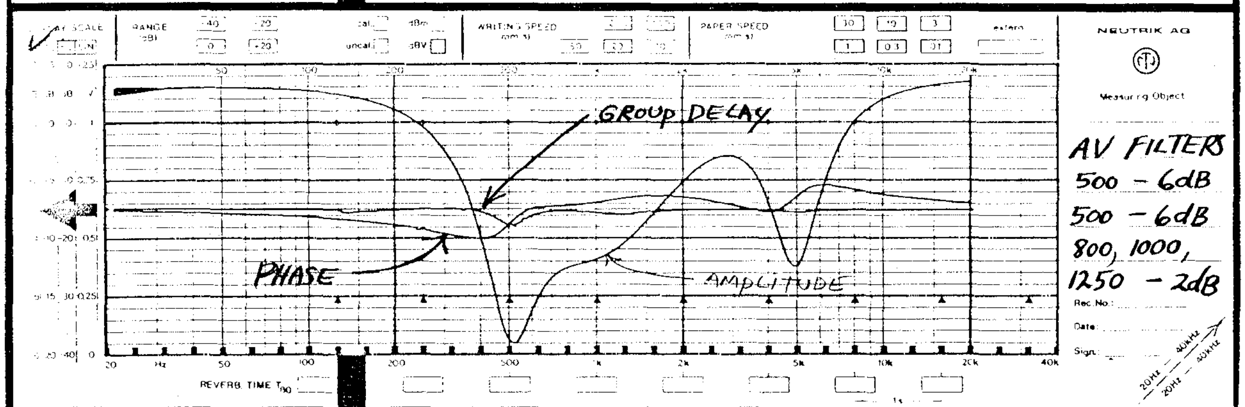
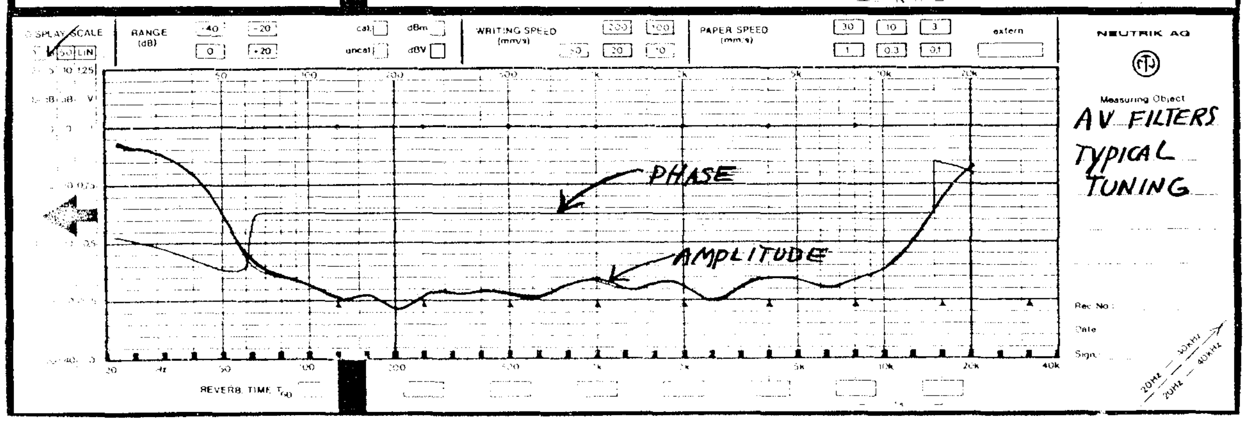
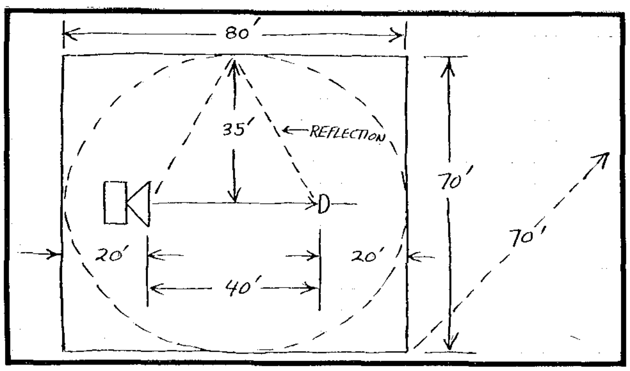






APPROXIMATE ROOM CONSTRAINTS ON 30Hz MEASUREMENTS



1. Microphone to Loudspeaker Distance slightly greater than one wave length.

2. Reflections all travel a length *difference* as compared to the direct sound of slightly greater than one wave length.

"NEVER PAY ANY ATTENTION TO WHAT A THEORETICAL PHYSICIST SAYS, WATCH WHAT HE DOES"

Einstein once said, "Never pay any attention to what a theoretical physicist says, watch what he does."

I was reminded of this as I read the interview with Amar Bose in the July 1983 *Audio*.


Amar Bose

Fritz Winckel has written in *Music, Sound, and Sensation*, "In music intonation and color formation obey aesthetic laws of deviation from strict harmony and perfect form." Syn-Aud-Con says if you can't hear it, why measure it? Bose is quoted as saying that he set out to find out why "something that measured well sounded poor."

Later in the article Bose states, "My formal field of research was more in electronics and *communications theory*."

The one hint Bose was willing to drop in this interview is that they possess *software* "to make measurements which encompass those elements *that we know about* and *can handle* concerning how a human processes sound to make his final judgment on it" and then he adds "and that's highly proprietary."

It should be obvious to the most casual observer that the Bose company has their act together and that it is *different* from the majority of other manufacturers in the same market place. That market place is, in reality, the user seeking pleasant music creation in his own home, not recreation of a physical event at some other site.

We certainly agree with Bose when he states that "What is in the textbooks (at the time period of which he speaks) as to how to design loudspeakers is absolutely incorrect. Heyser has proven this in detail and his adaptation of Dennis Gabor's (communication theory) analytic equations to this purpose has been a rare technical "tour de force."

Syn-Aud-Con is firmly of the opinion that all psychoacoustic effects theoretically can be measured. What's wrong with "measurements" is that, as the TEF® analyzer clearly reveals, the wrong measurements have often been presented in the past as immutable truths. Hooray for the skeptics who challenge such assumptions.

WE NEED INSTALLATION PICTURES NOW

We are re-writing the yellow book NOW.

We should have put a notice in the Newsletter several issues back asking for outstanding installation pictures illustrating good design and innovation.

While we already have given 7 chapters to the publisher, in some cases pictures were not supplied, knowing that this does not affect their work on the galleys. We can add pictures by making a figure 1a, 1b, etc. or we can substitute pictures we are now using.

If you have pictures of a sound system installation, in fact, anything (an idea, an installation, etc.) you feel would be a useful addition to the book, please send immediately, with all credits attached (architect, consultants) as well as a short description. We'll be sure proper credit is given for each picture used.

We need an example of: (1) Pew Back; (2) single source cluster; (3) single source; (4) distributed in-line array; (5) distributed circumferential arrays; (6) heavy coverage overhead ceiling distributed systems; (7) indoor and outdoor systems; and (7) anything we've missed.

Photos not used in the book will be returned to you.

RICHARD C. HEYSER ON "GROUP DELAY"

The following is excerpted from Pichard C. Heyser's article, "Concepts in the Frequency and Time Domain Response of Loudspeakers," *Monitor* - Proc. IREE March 1976.

From page 69:

"How can we deal with this delay? Our first reaction is to turn to the use of group delay {16, 17} but when we do that we immediately encounter logical inconsistencies. A headlong application of group delay to loudspeaker measurement discloses that for many frequencies the pressure wave is calculated to occur before the electrical signal giving rise to it! This violation of causality is absolutely unacceptable. Group delay clearly cannot be used. The reason for this is that group delay, defined as the negative of the rate of change of phase with frequency, cannot be used as a measure of time delay when the amplitude response shows a non-zero rate of change with frequency. In fact, it has recently been demonstrated that group delay can never be used as a measure of time delay when applied to any non-trivial minimum-phase network. {10} Exit group delay. Then what is left?

"The solution to this problem must be offered by a completely new look at network time delay. Without going into details since it is covered elsewhere, {10, 18} there is one special type of network for which the time delay and group delay coincide. In this network the time delay of each frequency component is always positive and coincides with the definition of group delay corresponding to the negative of the slope of the phase versus frequency. The network is the all-pass transmission function which is strictly non-minimum phase.

"What has this got to do with loudspeakers and other networks which have a frequency-dependent amplitude response? The surprising answer is that properly mixed parallel combinations of all-pass networks can be used to imitate any reasonable minimum-phase response function, regardless of the variations of its amplitude with frequency.

"This means that since we know the true time delay properties of the all-pass networks and since signal components can be linearly added if we have a non-interacting parallel combination of such all-pass networks, there is a proper analytical tool that can be used to calculate the amount of time delay that can be expected from a loudspeaker or any network element.

"If a signal is fed to a loudspeaker, the resultant response can be equated to that of an ensemble of parallel all-pass elements. Each all-pass element has a perfectly uniform amplitude response and a frequency-dependent phase response with a slope that never becomes positive with increasing frequency.

"A dip in the loudspeaker frequency response corresponds to a cancellation by destructive interference in the equivalent ensemble response of all-pass elements. A peak is the converse. Since this is a model, we know that the actual loudspeaker, as a network, is not composed of these conceptual elements but the fact remains that if we compute and measure as though it were made of these elements, we will achieve results indistinguishable from the physical case.

In conclusion of the article, pages 74-75:

"What I actually found was that I could learn from the loudspeaker how to improve the holes in my own background of communication theory. A case in point is group delay, a concept which I frankly never understood in college studies but which I had been assured was properly defined "somewhere." Everyone knows that there is some actual time delay for a signal passing through a physical system but when group delay, which pretended to be that delay, was applied, non-causal solutions frequently resulted. Textbook derivations made a great deal of fuss about explaining that "anticipatory transients" did not really exist and could be explained as resulting from approximations in the presumed frequency response. It was vaguely hinted that group delay could not be used near frequencies of absorption. But how close? And was it valid farther away? If so, then how far away?

"As long as such problems lurked only at the fringes of measurement, no-one seemed bothered. But suddenly with loudspeakers, I found myself deep in such considerations. The problems could not be ignored, particularly since they seemed to be at the heart of considerations of the importance of phase response to quality of reproduction.

"I did finally find a solution to the mystery of time delay in a dispersive absorptive medium but only after I was forced to do so by practical considerations of loudspeaker reproduction. This solution has since been folded back into electromagnetic problems to obtain correct answers. Group delay, it turns out, is never equal to causal time delay in a minimum-phase system. One should not compute the time delay of a loudspeaker by a measurement of group delay."

J. W. DAVIS SBA INTERCHANGE

Ken Wahrenbrock is the editor of a Newsletter for J. W. Davis & Company and will spread the word about the SBA system. To help this program we want to offer a free subscription for one year to anyone sending in usable photos of an SBA installation.

Let us hear from you.

HEYSER PAPERS

We have prepared a list of all the Richard Heyser papers we can locate so we can have them all in a manual of Heyser papers.

If anyone knows of any papers we have not listed, please let us know. We showed the list to Heyser several years ago and he thought the list was complete, but we would like to be sure.

	No. of Pages	Title
1.	14	Acoustical Measurements by TDS - AES 10/1967
2.	12	Loudspeaker Phase & Time Delay Distortion Part I - JAES 1/1969
3.	8	Loudspeaker Phase & Time Delay Distortion Part II - JAES 4/1969
4.	10	Determination of Loudspeaker Signal Arrival Time Part I - Vol. 19, #9 JAES 10/1971
5.	6	Determination of Loudspeaker Signal Arrival Time Part II - Vol. 19, #10 JAES 11/1971
6.	4	Determination of Loudspeaker Signal Arrival Time Part III - Vol. 19, #11 JAES 12/1971
7.	12	The Delay Plane, Objective Analysis of Subjective Properties Part I - Vol. 21, #9 JAES 11/1973
8.	6	The Delay Plane, Objective Analysis of Subjective Properties Part II - Vol. 21, #10 JAES 12/1973
9.	6	Breakthrough on Speaker Testing - Audio 11/1973
10.	1	Speaker Tests - Impedance - Audio 10/1974
11.	3	Speaker Tests - Phase Response - Audio 12/1974
12.	1	Speaker Tests - Room Test - Audio 1/1975
13.	3	Speaker Tests - Polar Response - Audio 5/1975
14.	6	Some New Audio Measurements - AES Preprint 1008 5/1975
15.	6	Geometry of Sound Perception - AES Preprint 1009 5/1975
16.	3	Some Useful Graphic Relationship - JAES 1976
17.	6	Holomorph Recording - AES Preprint 1115 3/1976
18.	10	Concepts in the Frequency and Time Domain Response of Loudspeakers - Monitor - PROC. IREE 3/1976 (Australian Publication)
19.	1	Square Wave Testing - Syn-Aud-Con Newsletter 1976
20.	2	Crescendo Test - Audio 5/1976
21.	4	Time & Frequency in Loudspeaker Measurements - Audio 7/1977
22.	3	Fuzzy Alternatives - JAES, Vol. 26, #3, 3/1978
23.	3	Hearing vs. Measurement - Audio 3/1978
24.	8	Imprecise Descriptions - AES Preprint 5/1978
25.	3	Acoustic Rosetta Stone - Audio 1/1979
26.	5	A View Through Different Windows - Audio 2/1979
27.	6	Catastrophe Theory & Its Effect on Audio Part I - Audio 3/1979
28.	5	Catastrophe Theory & Its Effect on Audio Part II - Audio 4/1979
29.	9	Catastrophe Theory & Its Effect on Audio Part III - Audio 5/1979
30.	2	The Forum - Polarity Convention - Audio 9/1979
31.	1	The Concept of Distortion - Syn-Aud-Con Newsletter Summer/1980

SIGNAL DELAY vs SIGNAL TIME INTERVAL

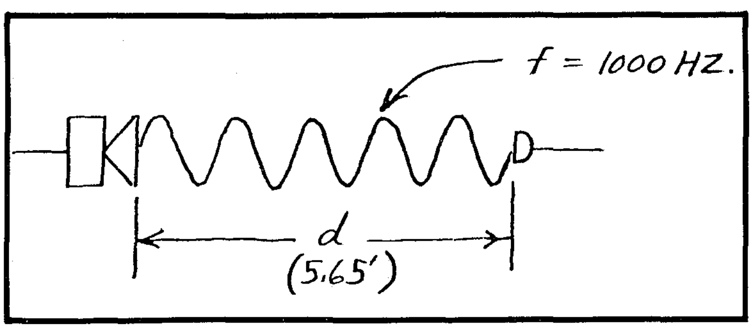
Signal Time Interval $= \frac{1}{f} = 0.001$ sec.

Signal Delay $= \frac{d}{\text{vel.}} = \frac{5.65}{1130} = 0.005$ sec.

Signal time interval is frequency dependent.

Signal delay may or may not be frequency dependent.

A frequency time curve (FTC) is the apparent variation in d (expressed as a time interval) as the frequency varies.



AUTOMATIC MIC MIXING FOR "LIVE" MUSIC

In March 1982, before Wolf Trap burned, Farrel Becker prepared a 60 page report "detailing the steps taken to improve the Federc Centre's sound system during 1981 and 1982.

Quoting from pages 16 - 20 of the report:

Several techniques, developed and refined in the past few years, were utilized during the performances of 1980 to reinforce the live sound. By far the most effective technique was the method utilized by the sound system operators in mixing the shows. This requires a great deal of talent and skill developed only with experience. The operators would have to control, typically, five microphones placed on the stage floor and spread evenly across the front of the stage. Occasionally, depending upon the nature of the scenery in use and the blocking, additional microphones would be placed in various locations on the set. The operation of this setup requires the operator to open, or turn on, whichever microphone is closest to the performer who is singing at any particular time. The volume of that microphone is adjusted so that the singer is heard as clearly as possible, but still sounding natural, and not so loud as to make the audience aware of the sound coming from the loudspeakers. As the performer moves across the stage, the operator must follow him by opening the next microphone and simultaneously closing the first. This must be done smoothly or a momentary drop in sound level may be detected by the audience or, worse, feedback may occur. This procedure may seem rather simple, but it requires a great deal of concentration on the part of the operator. He must listen intently while watching the performers and continuously adjust the controls on the mixing console as the performer moves about on stage. With several principals, and perhaps a chorus as well, the continuous manipulation of the controls becomes very complex.

There are two basic reasons for this constant opening and closing of microphones. One is due to the effects of inverse square law. We are all aware of the fact that the level of a sound decreases as we get farther away from its source. An approaching airplane becomes louder as it gets closer to us and decreases in level as it gets farther away. The level of sound is inversely proportional to the square of the distance to its source. This means that every time we double the distance from the source, the sound pressure drops by one half. Every time we reduce the distance to the source by one half, the sound pressure doubles. An increase in sound pressure by a factor of two or a decrease by one half is equivalent to a change in level of 6 dB. The human ear detects a 10 dB change as being either half or twice the original level, so a 6 dB change is quite noticeable. When a performer is standing 12 feet away from a microphone and then moves to 4 feet away from the microphone, the level of his voice will increase 9.5 dB. Suddenly this performer is heard through the sound system at twice the previous level. However, in moving the 8 feet required to bring him to within 4 feet of the microphone, he has moved from 100 feet away from a listener in the orchestra seating to 92 feet away, a change in level of only .7 dB. While the level from the sound system has essentially doubled, the natural sound for most listeners has increased by an amount that is just barely noticeable. The result is that all attention is focused on the sound system, now blaringly loud. The operator must continuously compensate for this by adjusting the gain of the microphones.

The other reason for constantly manipulating the controls has to do with the gain available before feedback. As discussed earlier, the performers are usually five to fifteen feet from the microphones. With these great distances between singer and microphone, all of the available gain is usually needed in order to achieve a high enough level of direct sound for good intelligibility. Since the location of the loudspeakers is fixed and the location of the microphones is dictated by the set and blocking, there remains only one factor that can be adjusted to maximize the amount of gain before feedback, that being the number of open microphones. Every time we double the number of open microphones we lose 3 dB of gain before feedback. This corresponds to a loss in level of approximately 30 percent. When we need all we can get, a 30 percent loss is intolerable. This is why the operator must continuously open and close microphones. He must turn off all those that he is not using while adjusting the balance between those that he is. As the action takes place all over the stage, the operator is making these adjustments quite rapidly.

An experiment was performed with a new type of mixer during the week that the New York City Opera was at Wolf Trap. This new mixer is one of several on the market known as "automatic mixers". These mixers internally monitor the number of open microphones and electronically adjust the overall gain to compensate and keep the volume steady. Although they are intended primarily for conference room use, the design of one of these mixers lent itself to a unique method of connection to Wolf Trap's (or any other) existing mixing console. With the automatic mixer installed and properly adjusted, a great deal of pressure was taken off of the operator. He could now operate the controls with greater ease, and, since the mixer was now monitoring the number of open microphones, he could obtain even more gain before feedback than had been possible under completely manual control. The operator still had to adjust the volume of each individual microphone in order to maintain naturalness and a proper balance between singers and orchestra. No machine can do this. This experiment, with a mixer on loan from the manufacturer, was a great success. It was the first time that an automatic mixer had been used in such a way for the reinforcement of an opera. The purchase of an automatic mixer was recommended as a priority item.

To those unfamiliar with the complexities of sound reinforcement in the real world, it may seem advantageous to set a level for the microphones and then make no further adjustments in the interest of naturalness. As we have just seen, this simply is not possible. It is said by some that the constant "twiddling" of knobs is what ruins sound. It is the IMPROPER operation of these controls that produces an inferior sound. The operator MUST constantly adjust the controls in order to have natural sound.

Continued on next page . . .

And quoting from a news release from outstanding consultants, Klepper, Marshall, King Associates:

Acoustical consultants, Klepper, Marshall, King Associates, recently experimented with an Industrial Research Products DE-4013A automatic microphone mixer for musical theatre sound reinforcement. The unit was employed at the Arie Crown Theatre, Chicago, for the four-week engagement of Gershwin's "Porgy and Bess", produced jointly by Radio City Music Hall and Sherwin M. Goldman Productions. Larry King, principal in KMK, decided to use the DE-4013A in this application for several reasons: 1) the full 100 ft. stage width of the Arie Crown was to be used for the large performing ensemble which required eight footlight microphones (in this case, AKG C-45a's with CK-1 cardioid capsules); 2) the Arie Crown's sound operator, Pat Reynolds, was to be responsible for operating two "thunder and wind" sound effects tape reproducers simultaneously during the hurricane scene, necessitating "extra hands" on the control console, and extra human hands were not available; 3) artistically, the producers wished to employ only a subtle boost for the singers' voices, as appropriate to the operatic nature of Gershwin's work.

In addition to the automated group of footlight microphones, three AKG C-451/CK-9 "shotgun" condenser microphones at the footlights were mixed in the conventional manner on the theatre's Yamaha PM-1000-32 console. The shotguns were used only for distant (10-25 feet) pickup, as required.

Interestingly, the "automatic" on-off switching and related gain adjustments could be detected easily when listening over headphones to the output of the DE-4013A, but these effects were completely masked when its amplified output was mixed acoustically with the theatre's direct and reverberant sound field.

KMK plans to include automated microphone mixers in future theatrical sound system designs as a "smart" electroacoustic tool for theatre sound operators.

Automatic microphone mixing for musical theatre sound reinforcement as pioneered by Farrel Becker will become more and more common now that innovative consultants such as KMK are using the idea.

COMMENTS ON HOPKINS-STRYKER TERMS

1. L_W should, when used, be the L_W of the devices supplying L_D to the site where L_P is to be calculated. It should never be the L_W of the total array. The N factor accounts for the ratio of array power to the device power.

2. *There are no Q multipliers* or modifiers. Q changes around a loudspeaker pattern with changing level because *Q is always a point* not an area. Single Q values are assumed to be "on axis." $Q_{(\angle)}$ is taken to be the Q at that indicated angle off axis.

3. M_e, M_a, N, etc., are critical distance D_C modifiers.

4. The direct sound level L_D is affected by: L_W, Q, D_X, M_e

5. The reverberant sound level L_R is affected by: L_W, $S\bar{a}$, M_a, N

6. Be careful to avoid fundamental errors such as the R' advocated by some. R' fails to provide the modifier to the M_a factor that corrects for the difference in area covered and the ratio of the device's power allotted to that area.

$$M_a = \left(\frac{1 - \bar{a}}{1 - a_C}\right)\left(\frac{\text{Q required to cover area}}{\text{Q of device used}}\right)$$

   Since John Prohs' sphere mapping makes it easy to obtain the ideal Q that would cover the audience area, M_a can be obtained to a useful accuracy by such means.

7. Where true L_R exists, do not adjust L_D by changing the L_W of a device but rather by changing the Q of the device (i.e., the near throw horn should be 1/4 the Q of the far throw horn). In free field situations, the L_W can be varied. In the L_R situation, varying L_W leaves the device with the highest L_W in total control of L_R.

8. The Sabine equation is the most accurate in truly reverberant rooms. None of the equations are accurate in "dead" rooms. Thus, when using the Hopkins-Stryker equation in reverberant rooms, use $S\bar{a}$ and M_a not R (the room constant).

9. Remember that in the D_C equations we combine all factors for computational convenience. *You* must remember which sound fields they affect.

$$D_C = 0.141\sqrt{\frac{Q\ S\bar{a}\ M_a\ M_e}{N}}$$

10. There is a recent tendency to rewrite these conventional equations so as to elicit new constants and then discuss them as if the writer derived them. Learn to recognize the origins and only adopt the deviations when they demonstrate utility in the application rather than deviant thought in the composer.

SEATTLE - MAY 20-21, 1983

SAN DIEGO - JUNE 3-4, 1983

HP-41C CLUB REACTIVATED

Mark and Peggy Laffin of TaBoo Custom Design in Tucson have negotiated with John Lanphere to take over the task of receiving, proofing, and distributing the HP 41C subroutines submitted by Syn-Aud-Con graduates. John Lanphere states that he simply got swamped by material sent in. (It is far more difficult to proof programs than it is to write them from scratch.) Mark and Peggy operate a business specializing in HP 41C software (Mark's main occupation is as an engineer at Hughes) and they feel they have both the experience and the resources to support this sharing of programs. The fact that they must charge more than John asked is evidence that they have the experience.

A major problem with this work is that the programing may be faultless but the program itself is technically incorrect in the acoustical sense (i.e., the L_W of the array is used rather than the L_W supplying L_D or Q multipliers rather than D_C modifiers and other misunderstood principles that can "hide" deep in a superbly written and documented program).

Syn-Aud-Con approves of sharing programs but our basic philosophy should be adhered to if you want to avoid serious sound system design flaws; namely, understand yourself exactly what each equation in the program is doing and what if any are the limiting constraints on its use.

1. The annual club fee is $15.

2. TaBoo expects that members will support the club in terms of programs. To submit programs, members will be required to include complete documentation. Undocumented software will be returned to the author.

3. TaBoo will make available for a small fee, a documentation package similar to that required by HP for the HP user's library.

4. There will be a dedicated committee of editors to review programs. Editors will be responsible for putting submitted software through its paces. They will make corrections and suggestions where necessary. TaBoo will be responsible for putting submitted programs in their finished form and distributing them.

5. TaBoo will publish a quarterly newsletter containing the titles, descriptions, and prices of programs currently available. The newsletter will also contain a user feedback section to offer members a place to make suggestions or corrections. A classified section may also be included.

TaBoo expects to make the first mailing 45 days after John Lanphere has turned all material over to them (which should have been done by October 1st). The mailing will be made to names given them by John plus any new interested people. So, if you are interested, contact TaBoo Custom Designs, P. O. Box 37017, Tucson, AZ 85740.

Let TaBoo know if you would be willing to serve on the editorial staff. They would like a sufficient number of editors so that each editor would "proof" only 1 or 2 programs a year. We hope to have several editors from abroad. If you have any suggestions for the club, contact TaBoo.

HP-41C SOUND SYSTEM DESIGN PROGRAMS

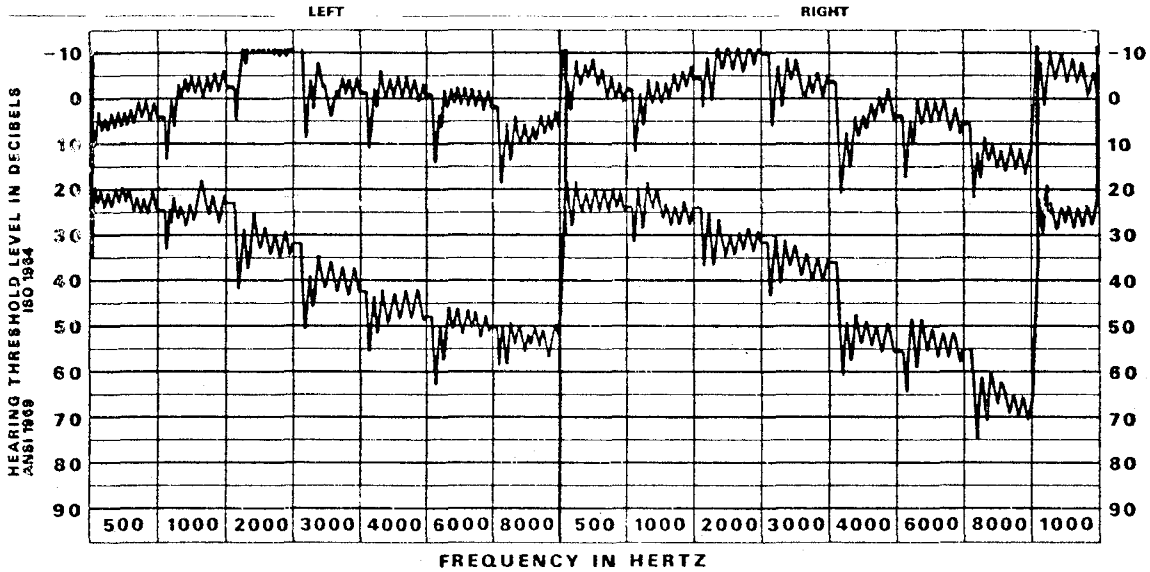
One of the most complete programs on sound system design has been written by Jeff Long and will be one of the early programs sent out for review. (Don spotted a few changes he would like to see made.)

A letter from Johan v. d. Werff in the Netherlands illustrates the eagerness and hunger for audio communication. Johan told us that he got in his car and drove all over The Netherlands asking every audio and electronics shop owner what they knew about $\%AL_{cons}$. He finally found someone who said that he personally didn't know anything about it but he had seen a "yellow book" .... Johan then learned that the man who knew more about articulation loss of consonants lived only a few miles away in Nijmegen.

Johan is seen here with Don Eger at the AES in Eindhoven.

HEARING

The upper trace is an audiogram of Carolyn taken a few years ago. The lower trace is taken with E.A.R. protectors. It isn't a rigorous measurement, but it does give an idea of the E.A.R. protector - 40 dB through much of the higher frequencies.



Interesting Quotes

From *Audio*, December 1982, "The Only Component You *Can't* Replace: Ears" by Simeon Costa:

The Mark I ear--the kind you are currently wearing--is equipped with some built-in, although limited, protection against thunderous noises. Any loud, sudden bang causes a reflex action in two tiny muscles located in the middle ear. The *tensor tympani* contracts and stiffens the eardrum so it can't vibrate as freely as it normally does. And the *stapedius* muscle pulls on the little stapes (stirrup-shaped) bone which more or less immobilizes the movement of the little chain of bones that transmits sound from the eardrum to the inner ear. The overall effect is to reduce the transmission of sound and shield the delicate nerves of the inner ear.

This muscle-reflex action, while a kind of Rube Goldberg engineering, probably gave adequate protection in quiet prehistoric times. Thunderclaps are relatively infrequent and come at longish intervals, so the little ear muscles received an opportunity to rest between bangs. Today, however, things like engines and rivet guns bang, bang, bang hundreds of times a minute and for long periods. The tiny *tensor tympani* and *stapedius* muscles become tuckered out in a few moments and leave the ear wide open to a flood of destructive decibels.

(Editor's Note: Drugs and alcohol also make the *tensor tympani and stapedius* unresponsive "and wide open... to destructive decibels."

From *Sound and Vibration*, December 1982 "Effects of Noise on Perception of Speech by Children and Certain Handicapped Individuals" by Lois L. Elliott:

In summary, a number of population groups have difficulty understanding speech when their performance is compared to that of young, normal adults. These population groups include normal-hearing children, normal-hearing older adults, normal-hearing children with speech disorders, normal-hearing learning-disabled persons, and people of all ages with hearing-impairments. Quite young children have poorer-than-adult-levels-of-performance when listening at *low levels, in quiet,* to words that are well within their receptive vocabularies. All of these special groups exhibit some type of difficulty for certain types of speech materials and when listening in noisy or reverberant situations. Although considerably more research is needed to understand the reasons for these performance differences, it is important to emphasize that the signal-to-noise ratios and reverberation times used in these experiments resemble situations that are frequently encountered in real life. When designing structures for living and communicating, architectural acousticians need to attend to the special needs of these groups, which constitute a large proportion of the general population. When the noise sources are outside buildings, such as highway noise, or outside rooms where these persons work or play, appropriate noise-insulating approaches are needed. When these sounds are *within* the same living space - such as dishwashers and TVs in homes or scraping chairs and conversation of activity groups in classrooms, other quieting procedures are needed. In some instances the need is to design equipment (e.g. air conditioners) that is quieter. In almost all circumstances, sound absorptive materials are required. Most importantly, acousticians as well as 'consumers' (e.g. school superintendents, school principals, teachers, parents, retirement and nursing home directors, and municipal officials) must recognize that certain population groups require noise control measures to a much greater extent than do other groups if they are to understand speech - and decision-makers must take appropriate actions.

Continued next page.....

HEARING (continued)

From *U.S. News & World Report*, October 18, 1982, Interview with Dr. Ralph Nauton, Expert on Ear Problems, "Hearing Loss: Ways to Avoid, New Ways to Treat":

Q, Is it dangerous to listen to hi-fi music at full blast?

A From experiments with monkeys, we know that exposure to disco noise over a protracted period of time results in damage to the auditory nerve of those monkeys. It's most likely that with teenagers who are habitual discogoers and who don't protect their ears, there will be some subtle damage.

We're also concerned about the growing trend of walking and jogging with earphones listening to music. It's far easier to produce loud sounds under an earphone than with loudspeakers at a disco.

Q. Can some drugs cause hearing loss?

A A number of antibiotics--such as streptomycin--will damage parts of the ear, but these drugs are not usually prescribed except in hospitals for life-threatening situations.

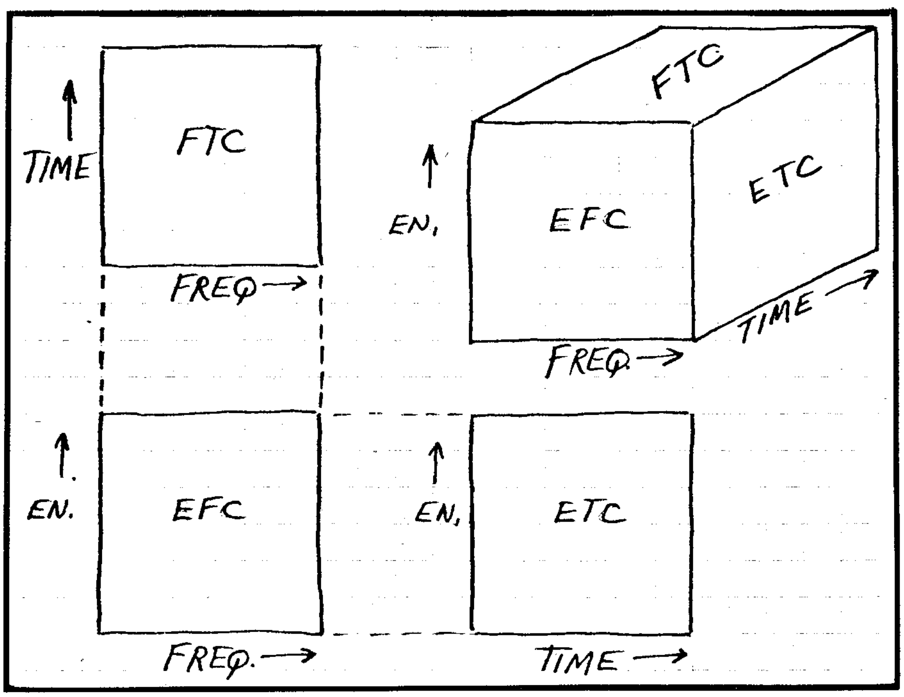
From *Time*, September 19, 1983, "Help for Reagan's Hearing":

Ronald Reagan, 72, has had a hearing problem for more than 40 years. He traces it to an incident during the filming of a movie around 1940 in which a fellow actor accidentally 'shot a gun off in my ear that almost knocked me down.'

and further on:

It was the availability of this new model, introduced last year, that led Reagan's otologist, Dr. John William House, to suggest that he try an aid. House, of the world-renowned House Ear Institute of Los Angeles, arranged for the President to be custom-fitted by Starkey Labs, Inc. in Minneapolis, one of a number of companies that makes the unit. The mini-aids are tuned during the manufacturing process so that they amplify the frequencies the wearer has trouble hearing; some 50,000 different settings are possible. As a result, Reagan not only hears sounds more loudly but more clearly, without the confusing magnification of background noise that has annoyed aid wearers in the past.

THE THREE FACES OF TEF® MEASUREMENTS



EFC is time blind

ETC is frequency blind

FTC is energy blind

3-D is slightly distorted view of all three

TEF® INSTRUMENTATION WORKSHOP

WHERE: ATLANTA, GEORGIA

WHEN: NOVEMBER 1-3, 1983

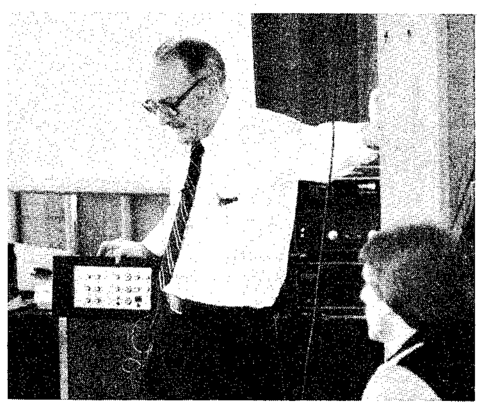
WHO: DR. EUGENE PATRONIS, Workshop Chairman - Physics Department, Georgia Tech

The acceleration of TEF® technology since the first delivery of the Tecron TEF® System 10 is staggering, even to those of us following every step. We are very pleased to be able to offer a super opportunity to interface with this technology guided by Dr. Eugene Patronis in a special workshop to be held in Atlanta, Georgia. He has unique qualifications:

- Dr. Patronis is fully conversant with all the mathematics bearing on advanced communications theory.
- Dr. Patronis is a very successful designer of practical sound systems of the most complex types.
- Dr. Patronis is the possessor of one of the very first TEF® analyzers and has at his disposal the very best of the conventional FFT analysis systems which he has critically compared to the TEF® method.
- Dr. Patronis is an enthusiastic teacher with a gifted ability to translate very difficult mathematical concepts into delightful visual explanations that leave the observer with a conceptual grasp of the underlying physical principles involved. One only has to watch Dr. Patronis playing the part of a sound wave traveling down a plane wave tube to appreciate that the theater lost a great talent when he took up physics.

This special TEF® workshop will feature abundant "hands on" opportunities at the keyboard of the TEF® analyzer.

Upon reflection, one of the key lessons from 30 plus years of experience in audio and acoustics is not to place yourself under inferior instructors when attempting to learn something. While it is a Syn-Aud-Con axiom that all true learning is individual and outside the classroom, the role of an inspired teacher in motivation, direction, and ready reference is incalculable. That's what is so truly unique about Syn-Aud-Con's special workshops. Those of you who have suffered through incompetents at technical society workshops, or had professors who raped their student's creative efforts to further their own careers, or those manufacturer's seminars wherein a powerful ego runs unrestrained by a critical peer group, can appreciate the sheer joy of studying a current technology under instructors who can provide motivation from a simple review of their own genuine accomplishments -- direction that allows you to meaningfully pursue a path free of dead ends and of such a sharing nature that they become invaluable reference sources for the remainder of your career. That's what these workshops provide. All you have to do is want to experience it.



Workshop Chairman, Dr. Patronis, with his right hand on a new "baby."

SYN-AUD-CON FALL SEMINAR SCHEDULE

September 13-14, 1983
St. Louis, Missouri

September 29-30, 1983
Chicago, Illinois

October 6-7, 1983
New York, New York

October 18-19, 1983
Washington, D.C.

October 26-27, 1983
Atlanta, Georgia

November 7-8, 1983
Orlando, Florida

November 15-16, 1983
Dallas, Texas

November 21-22, 1983
San Antonio, Texas

November 29-30, 1983
Houston, Texas

December 13-14, 1983
Las Vegas, Nevada

January 18-19, 1984
Anaheim, California

LEDE™ & STUDIO DESIGN WORKSHOP

WHERE: DALLAS COMMUNICATIONS COMPLEX @ LAS COLINAS

WHEN: DECEMBER 5-7, 1983

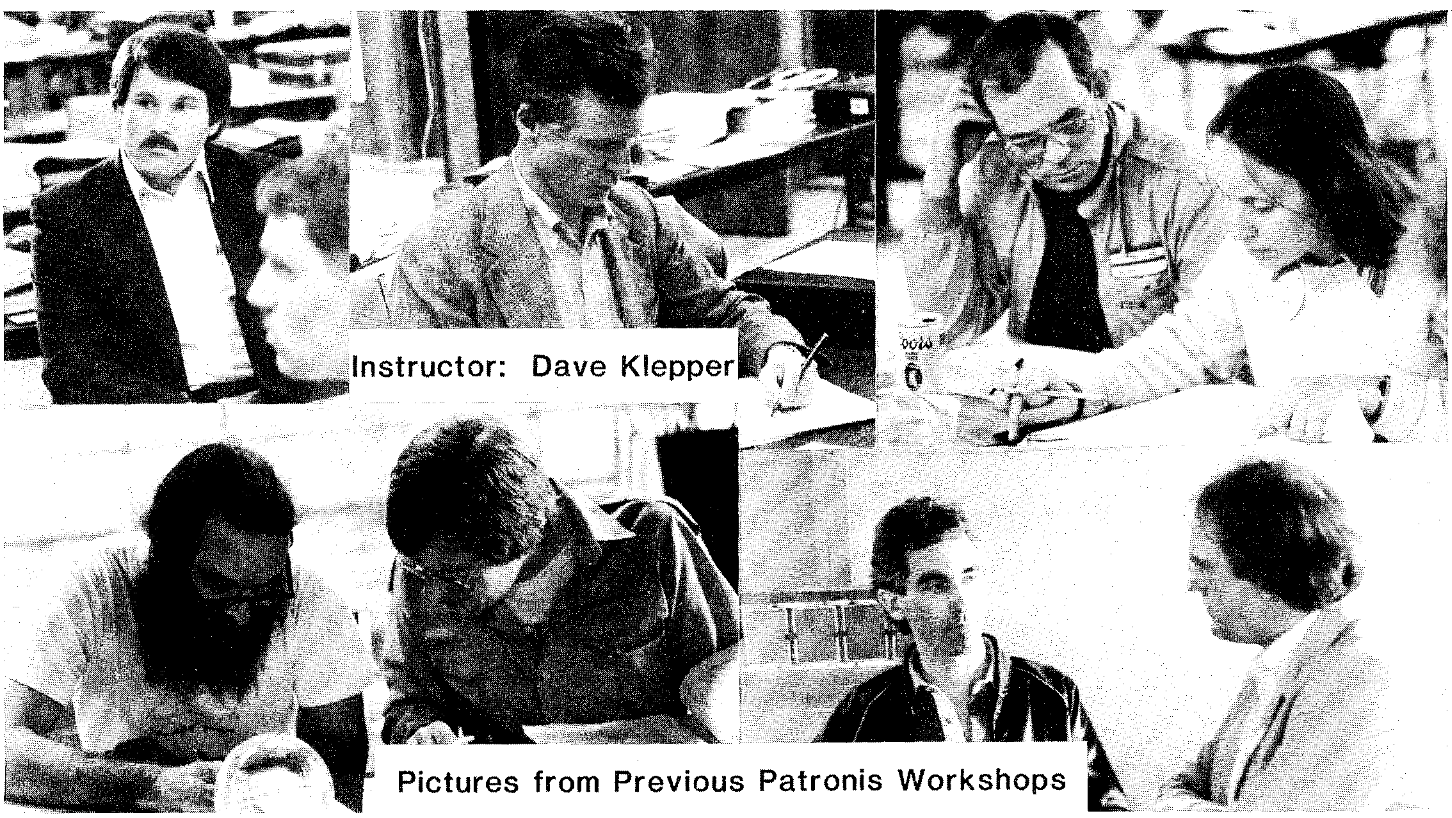
WHO: RUSSELL E. BERGER, Workshop Chairman - Assisted by the Staff of Joiner, Pelton, Rose, Acoustical Consultants

There are two very active builders of L.E.D.E. control rooms who have completed a sufficient number of these spaces to qualify as "experienced" LEDE consultants. One, of course, is Chips Davis. The other is Russ Berger of Dallas, who is a senior consultant with the firm of Joiner, Pelton, Rose.

Russ was among the very first TEF® licensees and already has on hand a TEF® Tecron System 10 analyzer which is being used in the evaluation and adjustment of his current jobs. Russ also is a genuine expert with his IBM personal computer and his programs for room modes, predicting ETC plots, transmission loss, and other exceptionally well-thought-out acoustic subroutines are in the hands of a number of Syn-Aud-Con graduates, thanks to Russ' generosity in sharing his programming.

Joiner, Pelton, Rose is just completing a new communications center on the north side of Dallas which includes a truly "state-of-the-art" LEDE control room at Dallas Sound Lab designed by Russ.

During our visit to this new facility in July we were so impressed with what we saw and heard that we asked Russ and Joiner, Pelton, Rose if they would be interested in working with us on the second Syn-Aud-Con LEDE and Studio Design Workshop. As the prospectus below reveals, they are prepared to share the best thinking of their principal consultants. This will be a fully documented presentation. We believe this staff has no limitations in answering any LEDE and studio design question you wish to bring to the workshop.



Instructor: Dave Klepper

Pictures from Previous Patronis Workshops

BASIC SIGNAL DELAY SYSTEM EQUATIONS

Signal Delay in msec =

$0.885^{*}(D_O - D_S - D_2) + HE^{**}(\text{msec})$

*Constant found by

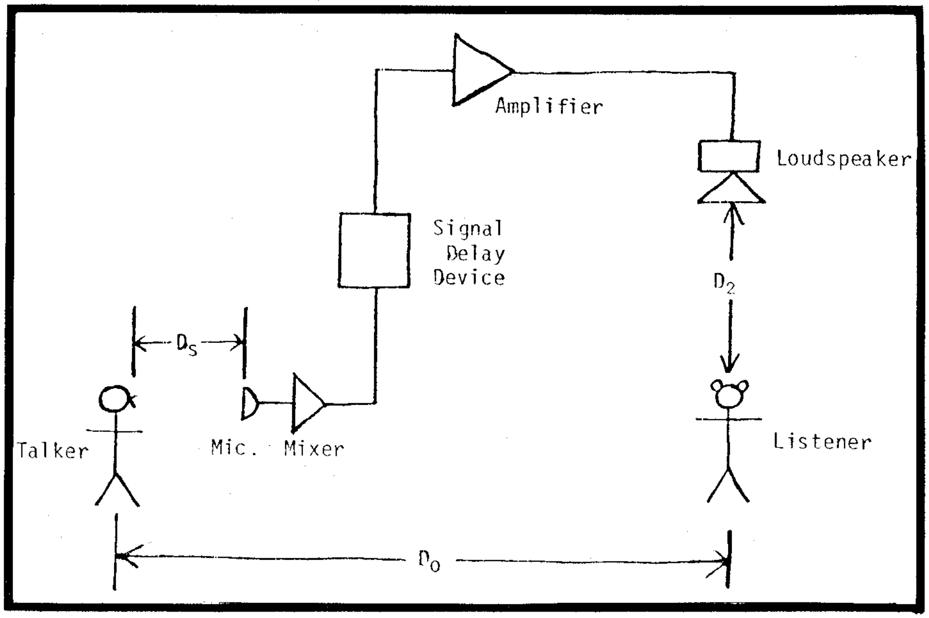
$$1000\left(\frac{1}{\text{vel. of sound (ft/sec)}}\right)$$

i.e.,

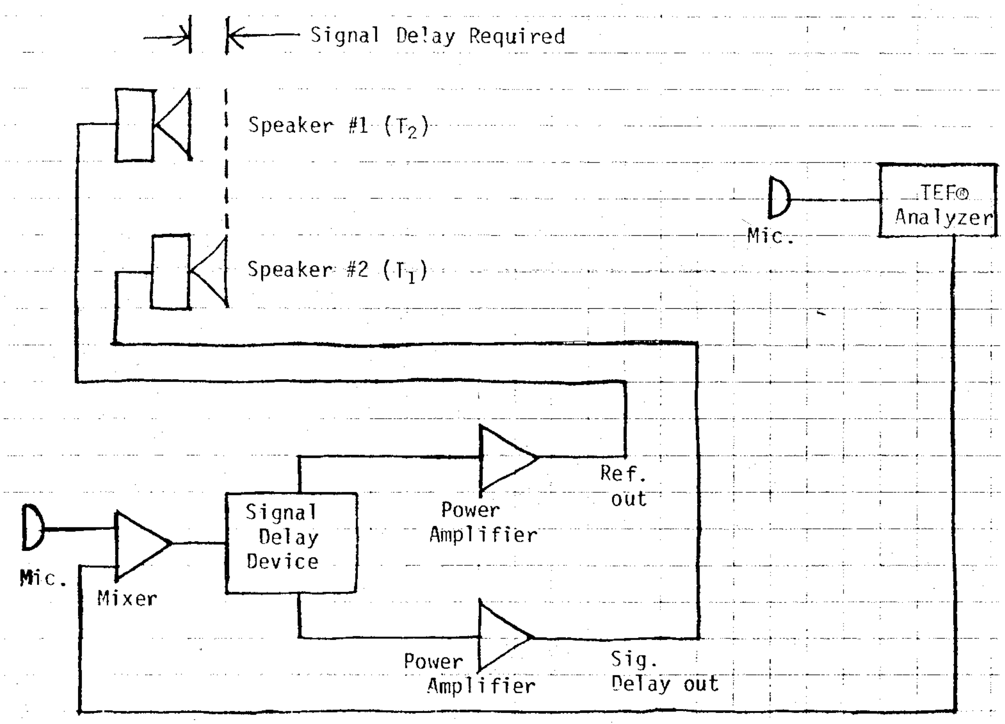
$$1000\left(\frac{1}{1130}\right) = 0.885$$

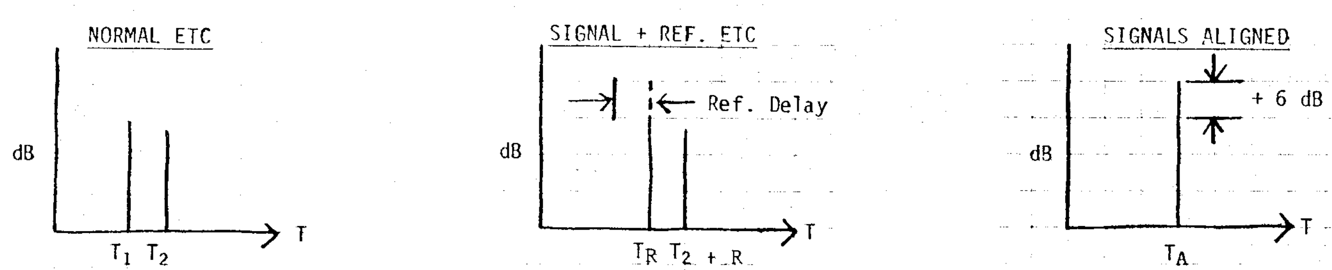
**HE (Haas effect) is the additional signal delay providing directional reorientation (usually 20 msec).

D_S was added to the equation for those instances where the microphone is worked several feet away such as in "live" theater productions.



SIGNAL ALIGNMENT DELAY DEVICES





A MODERN USAGE OF VOLTAGE REFERENCES

The two papers reproduced here by Jim Brown and Allen Burdick reflect the best practices of circuit designers today.

We can't help but comment on the dBu, however, if only in respectful memory of Mel Sprinkle. Please don't use any form of the dB in this manner. I would like to suggest that when the *voltage* 0.775 is used as a reference *amplitude* (never, never use the sacred term *level* for a voltage), let's call it 0 V_R (V_R means voltage reference). With this exception, I find these two letters helpful and informative.

From Allen Burdick, Benchmark Sound, Garland, Texas:

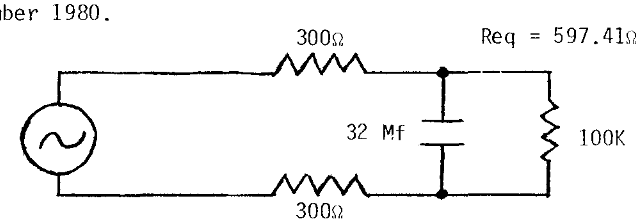
This letter is in response to the much needed Tech Topic by Jim Brown, Tech Topic Volume 10, No. 1, "Termination, Impedance Matching and the Maximum Power Transfer Theorem in Audio Systems," and is a point of further clarification. It is not uncommon these days to encounter lengths of cable of 1000 ft. or more between two pieces of equipment. Broadcast remotes in particular frequently see this much of an interconnect length. Jim points out that the distributed capacitance of a cable and the sending or source impedance from a single pole low pass filter may act as a RC filter or integrator degrading high frequency response.

Many of us, tutored under Walter Jung and Deane Jensen in proper circuit response and stabilities, can brag of flawless 10 KHz square wave response, with a system cut-off (not including power amplifiers) of over 100 KHz. No slewing induced distortion, no twin tone I.M. to be found in our systems. The local radio station is unable to trigger any pops or clicks here since we've done the proper and necessary stabilization. What a shock then, it is to find out what 1000 ft. of cable can do to the 600Ω system. The following chart shows the capacitances that I have measured on various types of cable.

Cable Brand	Type	Conductor to Conductor - C	Conditions of Measure
Belden	8412 Mic Cable	38.3 pf/ft	Over 10 ft
Mohawk	1742 Mic Cable	21.0 pf/ft	Over 10 ft
Belden	8451 Foil	31.7 pf/ft	*
Mohawk	1671 Foil	24.0 pf/ft	Over 10 ft
	1671 Foil	24.5 pf/ft	Over 1000 ft
Western Electric #22 2 Pair (station D wire)		16.0 pf/ft	Over 10 ft
Unknown Brand Telco Wire 3 Pair of #24		19.0 pf/ft	Over 10 ft

* "Audio cables" Reviews by Hugh Ford Studio Sound, November 1980.

The cutoff of any RC network is defined: $f_c = 1/2\pi RC$. In our case R is the parallel equivalent of the source and terminating resistances. C is the distributed cable capacitance and, of course, f_c is the cutoff frequency (-30 dB). Some quick math will show that with 1000 ft of cable @ 32 pf/ft, a source impedance of 600Ω and a bridging load of 100K:



f_c = 8.33 KHz ... is not exactly HI-FI. Terminating the line raises the cutoff to 16.6 KHz. This certainly is better in terms of response, however, we are back to the power matched system and the questions addressed by Jim Brown.

Well, what to do folks?

Richard Hess of ABC TV, in an excellent paper presented to AES in the fall of 1980, "Voltage Transmission for Audio Systems," AES preprint 1708, states that the results of his work indicate a 60Ω source impedance is optimum to drive today's foil shielded cables. One might expect from the above that 0Ω source impedance would be optimum. This is not the case, however, from two standpoints. First, some build out resistance is desirable to isolate load capacitance from the output of modern op amp circuits. This aids circuit stability.

Second, a lower source impedance than 60Ω actually produces some frequency response peaking due to inductance not accounted for in our simplified equivalent circuit.

Recalculating the cutoff frequency with a 60Ω source impedance, we find f_c to be 83.3 KHz -- a much better arrangement. If we deem 30 KHz as the lowest 3 dB cutoff frequency we can tolerate, then the maximum capacitive load is 88.42 nano-farads. This is the equivalent of 2947 ft. or approximately 900 meters of foil shielded cable at 30 pf/ft.

Continued next page.......

Another major advantage of the 60Ω source impedance is that there is only 0.8 dB difference in level between a loaded and an unloaded output versus the 6 dB level difference found in systems with a 600Ω source impedance. (Editor's Note: A 600Ω source terminated in 60Ω will be -6 dB. The author obviously intends to terminate??? 60Ω with 600Ω.) That 6 dB we used to throw away in build out resistors when an output was terminated is now available to us as additional output level (Editor: Misuse of the word level) or headroom if you will, and when an output is unterminated, there is an apparent 6 dB increase in gain. For instance, in the "wonderful" world of television the average audio level is a nominal +8 dBu, and with the 600Ω sourced system, the average unloaded output is a whopping +14 dBu with peaks as high as +28 dBu. Many a neophyte has been burned trying to connect a piece of gear with bridging inputs to such an output since most input stages clip at +22 dBu or less. That is the clip point of most currently available op amp circuitry. However, a few manufacturers are configuring the input stages of their equipment with a 6 dB loss to take advantage of the 6 dB of gain (Editor: Misuse of the term gain) (just discussed) intrinsic to the dual op-amp, low Z, balanced output circuit. This gives input and output clip points of +28 dBu, a healthy overload margin.

From Jim Brown, Bridgewater Custom Sound, Chicago, Illinois:

Thanks to Allen for his fine comments and for pointing out the excellent ABC papers from the October 1980 AES of which I was not aware.

I am in full agreement with the ABC papers and would encourage other broadcast and studio facilities to join the solid state age! Richard Hess's decision to standardize on a 60Ω source seems to be a good choice. I'd like to hear what some circuit people like Deane Jensen, Craig Connelley, and Bill Sacks think about it.

<u>Several other interesting points in the ABC paper:</u>

1. Hess's definitions of "Reference Level" and "Standard Operating Level" are also instructive.

"*Reference Level*" is the steady state setup level of the system corresponding to "0" on the VU meter or "-8" on the PPM. It is the indicated level to which we calibrate amplitudes using a tone.

"*Standard Output Level*" is defined as the indicated level at which distortion is specified for "clean" operation. (i.e., before it begins to rise at clip, or at transformer saturation.)

In more practical terms, it is the level at which we choose to operate our system on program peaks. To go higher yields unsatisfactory distortion; to operate lower drops us unnecessarily close to the noise level. Thinking in these terms, it becomes more obvious to us that what we need to start listening to is what things sound like at the *peak program levels* and not only at reference levels.

<u>Consider</u>:

The SMPTE I.M. distortion of a very good two-track, 1/4", 15 ips recording at 320 nw on Ampex 456 or Scotch 226 is approximately 1% AND IT DOUBLES FOR EACH 2dB OF LEVEL ON TAPE. Now, remembering that 320 nw is Reference Level and that program peaks are 10 dB higher, then the following table is instructive:

0 dB	=	320 nw	1% I.M.	+6 dB	=	640 nw	8% I.M.
+2 dB	=	403 nw	2% I.M.	+8 dB	=	804 nw	16% I.M.
+4 dB	=	508 nw	4% I.M.	+10 dB	=	1010 nw	32% I.M.

This distortion is inherent in the analog tape recording process.

That's a lot of distortion but since we're used to it, it doesn't bother us. But the guys in recording studios know it's cleaner at 30 ips because they listen to it every day and they can hear the difference!

Tape distortion is inversely proportional to the speed and also to the track width so a 1/2" 2-track tape has 1/2 the I.M. distortion of the 1/4" tape as does a 30 ips tape compared to 15 ips. Dubbing a recording yields 50% more distortion and 3 dB more noise.

Interestingly, each new tape developed and marketed by our tape manufacturers has 2 dB higher operating signal-to-noise level than the one below it in quality.

2. By using the PPM meter, ABC discovered that only 8 dB difference need be allowed between reference level (they use +8 dBm) and standard operating level (the level for specified distortion). If clip level is 6 dB higher yet, this means an output stage is needed which meets distortion spec to only +16 dBu and clips at +22 dBu. Before using PPM's, ABC had to allow 10 dB headroom, thus reducing headroom requirements 2 dB as a result of using the PPM's. That occurs because the operator is actually monitoring peaks.

Remember, peaks are what clip amplifiers, over-modulate tape and transmitters, and excite the F.C.C.

Hess also states that since standardizing on PPM's several years ago, "Operator acceptance at ABC has been better than anticipated. With our scale, program level is still run to '0' and the reference level setting is identified by the extended line at the '-8' marking, center scale."

Continued next page......

3. ABC-TV seems to have standardized on constant voltage distribution using 60Ω source impedance and 20K or higher loading (no termination). This combined with the use of PPM metering has reduced the level capability requirements of consoles, distribution amplifiers, and other peripheral devices from +30 dBm for a power matched system (150Ω source, 150Ω termination) to only +22VR for the constant voltage system. See Figures 1 and 2.

Most good consoles, distribution and line amplifiers, and other peripheral devices provide a clean +22 dBm to +24 dBm. Few will do a clean +30 dBm -- who needs it, really?

ABC has thus saved themselves the cost of a carload of line amps, 600Ω resistors, and the additional cost of making a distribution amp for +30 dB as opposed to +22 dBm. Last I looked, that's about 1/3 the cost of the DA! Not a bad savings when you need 25 to 100 DA's at $300 or so a copy. Not to mention the additional head load of the output stages and power supplies. Other advantages of constant voltage distribution as pointed out by Hess of ABC is that if multiple inputs are paralleled (accidentally or otherwise), no loss of level results and that since equipment is operating at lower power levels (and often with fewer line and distribution amplifiers), higher reliability results. There is less to go wrong.

4. The characteristic impedance of Belden 8451 cable is calculated by Hess as "in the neighborhood of 175Ω" although he makes no mention of having confirmed this by measurement. I wonder if this measured data is available for other commonly used cables and how it might relate to Harrison Klein's work on telephone line impedance measurements.

The ABC papers by Richard Hess and Hans Schmidt are excellent and should be required reading for anyone involved in audio systems engineering. (Editor's Note: The paper by Hans Schmidt referred to is *"The Peak Program Meter and the VU Meter in Broadcasting."* AES Preprint No. 1691 (1-8))

In Figures 1, 2 and 3 a typical modern line level output stage drives a typical line level input. The typical 20KΩ input stage is "bridging"; that is, it has an impedance high enough that it causes less than 0.5 dB loss (in this example, less than 0.25 dB in Fig. 1 and less than .03 dB in Fig. 2 and 3). *Since modern input stages are voltage operated devices*, what we care about is the *voltage* at the input terminals. Hence, the amplitudes are set so that in all three examples +8 VR program level (average level as read by a VU meter) appears at the input stage.

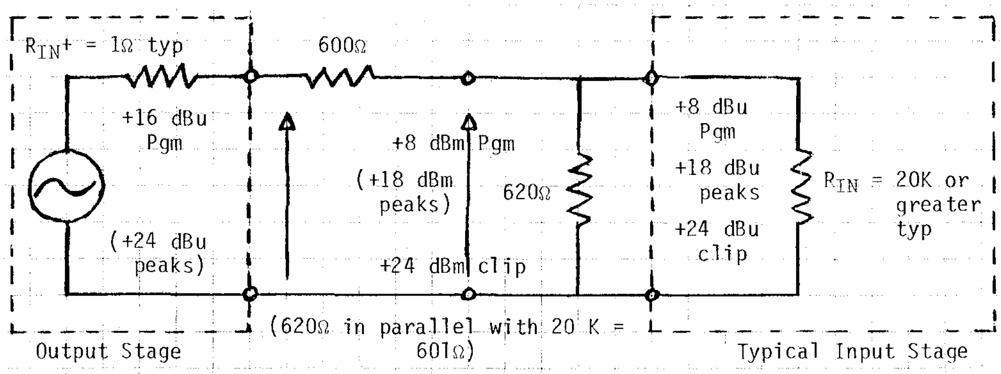


FIGURE 1: Power Matched System with VU Metering @ +8 dBm in 600Ω = +8 dBu

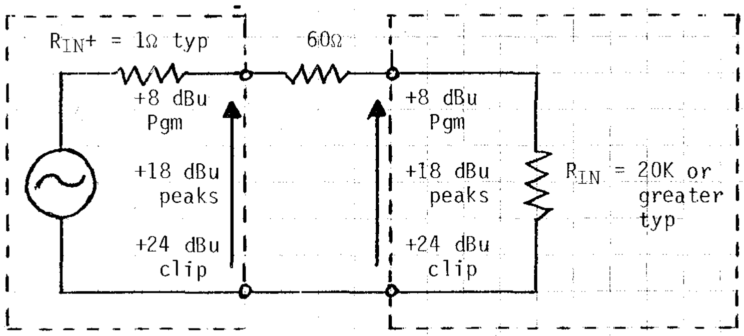


FIGURE 2: Constant Voltage System at +8 VR with VU Metering

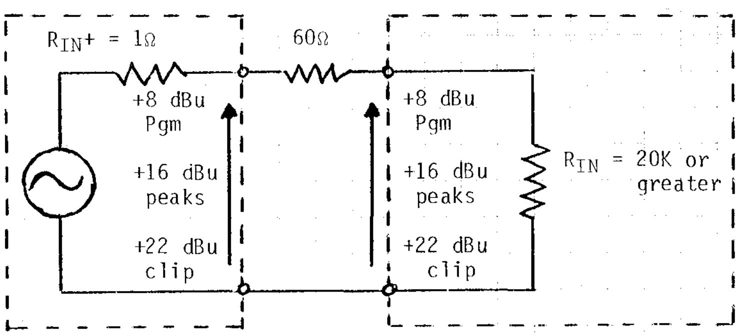


FIGURE 3: Constant Voltage System with PPM Metering @ +8 dBu

Note that in Figure 1, 6 dB is wasted across the voltage divider made up of the two 600Ω resistances whereas in Figures 2 and 3 less than .05 dB is lost in the 60Ω/20KΩ divider! Hence, the voltage swing required in the output stage of Figure 2 is only one fourth that of Figure 1 to get the same signal at the input stage.

Figure 3 shows the same relationships for PPM metering, where ABC has discovered that only 8 dB difference need be allowed between peak level and reference level, allowing an additional 20% reduction in output stage voltage swing. As Hess of ABC points out, these two changes reduce the *peak* voltage requirements of the output stage from ±17.5 volts for Figure 2 and ±13.8 volts for Figure 3.

In circuit terms, this means a simpler, less expensive, lower power (and lower distortion) output stage operating from a ±15 power supply may be used in Figure 3; ±18 V DC is required in Figure 2; and ±30 V for Figure 1. While an output transformer having a turns ratio providing some voltage step-up may be used to reduce the power supply requirements, this adds cost, size, weight and distortion.

MISUSE OF THE TERM VU

The term VU (volume unit) has a clearly defined meaning relative to program levels and a standard calibration technique relative to steady state sine wave signals. A 1,000 Hz steady state 0dBm sine wave signal is also 0VU. The reading in VU is the sum of the value read on the VI (volume indicator) meter and the associated sensitivity equipment (a passive attenuator in the classic case.) A modern meter, when indicating -4VU is actually 0VU because its sensitivity has been adjusted to match the older meters with their 3600Ω build out resistors installed in their attenuators, which requires a +4VU to be added to any apparent meter reading. (Note carefully that the meter indication *is not* the VU level.)

The term "VU" is often misused by the untrained as a *relative* voltage reference point which may be chosen at the whim of the user. Thus, we recently ran across the following table in a book on audio measurements.

"The industry standard levels that are used for '0VU' are as follows:

+8dBv (1.95 volts) = 0VU for the broadcast industry

+4dBv (1.23 volts) = 0VU for the recording industry

-10dBv (1.245 volts) = 0VU for home multitrack recording equipment

-15dBv (0.138 volts) = 0VU for musical instrument level equipment"

dBv (The dB with a lower case v stands for 0.775V - This same author acknowledges that there is a dB upper case V referenced to one volt.)

This is an appalling situation in which not one of the "Industry standards" is correctly described and, of course, there are no such standards. Let's discuss each area separately.

The Broadcaster

A broadcaster has *0VU = 0dBm for calibration purposes* and reads a *+8VU program level* as +18dBm (due to meter lag). The broadcaster, when at the +8VU level, reads an indicated 0 value on the meter and adds the attenuators +8 setting to it. The broadcaster is then at a level which requires a minimum of +18dB above 1 milliwatt power output capability *before* he or she adds +6dB of headroom and +6 dB for an impedance stabilizer pad. *(A total power requirement of +30dBm or 1 watt.)*

The Recording Industry

The problem here is that the recording "engineer" reads the meter only and to him or her an indicated "0" value on the meter means 0VU. Unfortunately, that is not true. However, if you can force all your friends and associates to accept that the VU has a new definition, namely that it is a relative voltage reference to be set as you please, then *partial* communication is possible with this *"engineer"* even if accurate *"level"* setting is not whenever you have one impedance and he has another.

Home Multitrack Recording Equipment

Here we have a real problem. First of all, the meter is not a VI meter, nor is it calibrated in VU, nor does it have proper ballistics characteristics. It is a device labelled to *"imitate"* a VI meter. Thus, its use in setting levels is about as useful as an imitation firearm in a gun fight.

Musical Instrument Level Equipment

A measurement session with much of this apparatus quickly reveals that specifications are on paper only. Why should they meet specs? Their competitors don't, their dealers wouldn't know the difference, and their customers will never check them. Even more reassuring, when the "tested in the home" gurus write reports on them in the trade magazines, they'll follow the "Industry standards" quoted above for their *"level"* measurements.

<u>WHAT CAN THE RATIONAL AND KNOWLEDGEABLE ENGINEER DO?</u>

1. Bask in the truth.
2. Seek out suppliers who know what they're doing.
3. Try to educate those suppliers worthy of saving for the future.
4. Avoid trying to "educate" those who are satisfied with a pretense of knowledge or who are simply rebellious about "good engineering practices."

<u>A FINAL WORD</u>

Remember that those who believe in these "Industry standards" do so out of simple ignorance of the existing real standards in use for the past 42 years. Help those interested in correcting their mistakes and let the others continue in their beliefs until they eventually discover the error of their ways the hard way.

READING "TRUE" LEVELS ON A VI METER

How to Read a VI Meter

Using a *Volume indicator* (VI) meter calibrated in *Volume units* (VU) across the output of a console often required calibration in order to read "true" levels.

A VI meter *is not* read from the meter itself. The meter is an indicator that should, whenever possible, be adjusted to show zero on its scale. The reading in VU is then the sum of the meter indication and the value indicated on the associated attenuator or the correction factor for the sensitivity, whichever happens to be the case at hand. For example, if there is a sensitivity adjustment, then applying a 1,000 Hz sine wave signal at a level of 0dBm to the line the meter is across will allow adjustment of the meter to an indication of -4 which when added to the sensitivity correction factor of +4 results in a 0 VU reading.

Let's place this meter in a typical situation, such as across the output of a mixer amplifier. The specified output Z is 600Ω. We measure it and find it is actually 110Ω. It is rated as having an output capability of +18dBm. What voltage will represent that level? Ok, but first what are we going to connect to it? Let's choose a power amplifier with a 15,000Ω input Z. The question then becomes if we generate an available input power (AIP) of +18dBm, what input voltage will appear across the 15,000Ω (i.e., what E_{IN})?

$$AIP = 10LOG\left[\frac{(E_S)^2}{.001\ R_S}\right] -6dB \qquad \text{Therefore:} \quad 10^{\left(\frac{AIP\ +\ 6dB}{10}\right)} = \frac{(E_S)^2}{.001\ R_S}$$

and

$$E_S = \sqrt{.001\ R_S\ 10^{\left(\frac{AIP\ +6}{10}\right)}} \qquad E_{IN} = E_S\left(\frac{R_{IN}}{R_S + R_{IN}}\right)$$

$$E_S = \sqrt{.001\ (110)\ \left[10^{\frac{(+18+6)}{10}}\right]} = 5.26\ V \qquad E_{IN} = 5.26\left(\frac{15,000}{110+ 15,000}\right) = 5.22\ V$$

Once this is established, any other level can be quickly calculated by:

$$+\ 18\ dBm\ to\ 0\ dBm = -18\ dB = 20\ LOG\left(\frac{XV}{5.22V}\right) = -18\ dB \qquad X = 5.22\left[10^{\left(\frac{-18}{20}\right)}\right] = .66V$$

Therefore, in this circuit 0 dBm is the level when .66V is read across the input of the amplifier. Now, let's suppose we plug a transformer into this amplifier's input and R_{IN} becomes 600Ω. What does E_{IN} become then for +18 dBm?

$$E_{IN} = E_S\left(\frac{R_{IN}}{R_S + R_{IN}}\right) = 5.26\left(\frac{600}{110 + 600}\right) = 4.45V$$

We would eventually find that if the mixer output were also made 600Ω, we would have exactly divided the E_S voltage of 5.26V to obtain E_{IN} (i.e., 2.63V).

CALIBRATING VOLUME INDICATING (VI) METERS

Standard Instrument

I. PERTINENT SPECIFICATIONS

- A. Meter Z = 3900Ω
- B. Attenuator Z = 3900Ω
- C. Build Out Resister (in Attenuator) = 3600Ω
- D. Total Z of Instrument as used = 7500Ω

II. SETTING CALIBRATION LEVELS

- A. Place Instrument across a 600Ω Circuit.
- B. Set Level of Circuit to 0 dBm at 1,000 Hz for a steady state *Sine Wave* Signal.
- C. With Attenuator set to lowest position (+4VU), adjust meter to read an *indicated* -4. Reading of true VU Level is 0VU.

III. HIGH IMPEDANCE VOLTMETER TYPE METER

- AA A. Place across circuit it is to be used in and measure the impedance of the circuit.
- B. Calculate E_{IN} for 0dBm at that point in circuit.
- C. Adjust meter to indicate -4 on that voltage.
- D. Remember to add +4 to any reading indicated on meter scale.

A UNIVERSAL dBm METER

Most meters labelled dBm meters are simple voltmeters. In order to read a *true level* they need to be across some specified reference impedance (usually 600Ω). But, zero dBm is one milliwatt (0.001 watt) and any voltage across any impedance that results in 0.001 watt *is 0 dBm*

$$\left(\frac{E^2}{R}\right) = 0.001\text{W} \equiv 0\text{ dBm}$$

Today's technology allows the economical construction of very "high gain stable" amplifiers. Such an amplifier with a very high impedance input could have its gain control calibrated for impedance mismatches allowing it to drive any standard voltmeter to a 0 dBm indication for any impedance labelled on its gain control.

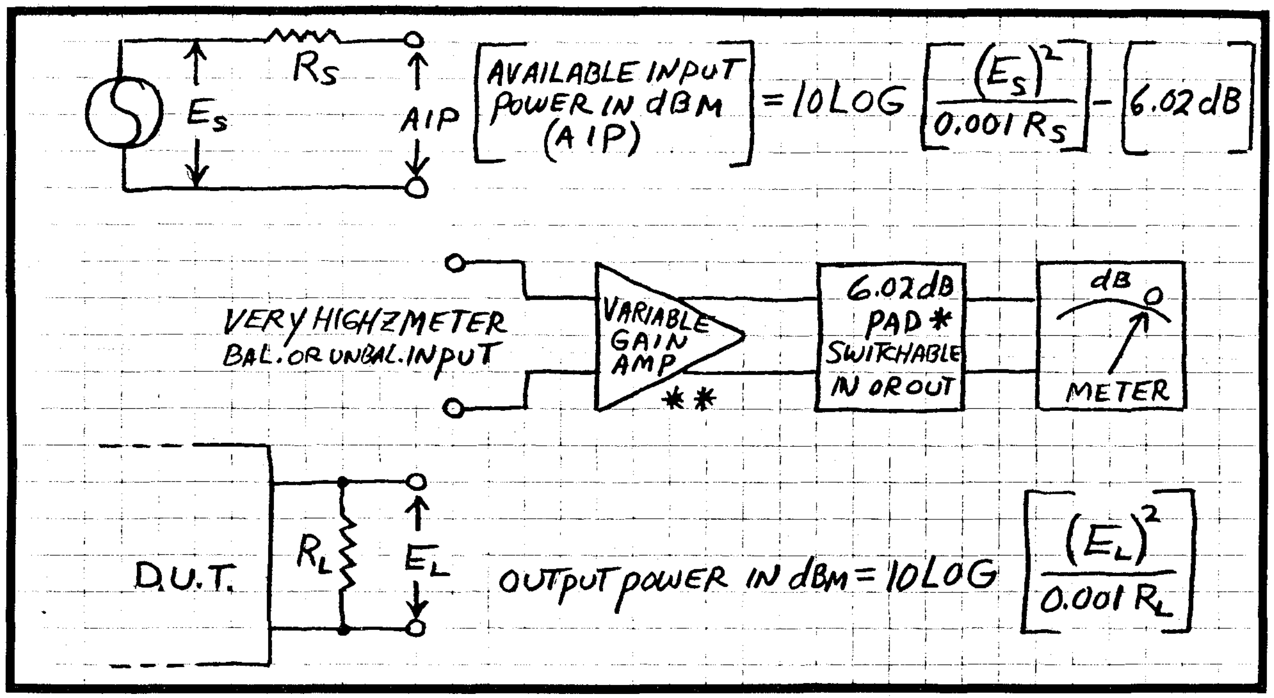
So long as the user remembers that the meter is indicating level and does not actually present the impedance labelled on the Gain Control, but only sees the R_L or R_S it is attached to such an amplifier should prove universally useful.

The gain for each labelled Z is found by:

$$\text{GAIN} = 10\text{LOG}\left(\frac{600}{Z}\right) \quad \text{or:} \quad Z \text{ equivalent} = \left(\frac{600}{10^{\left(\frac{\text{GAIN}}{10}\right)}}\right)$$

If the amplifier input is made to be a very high impedance switchable between balanced and unbalanced, almost any type of circuit can be measured.

The meter would have a dBm scale only. There would be, on the amplifier, the usual attenuator starting at any desired value (i.e., -60 all the way up to +60) in 10 dB steps.



* Switch controlling pad is labeled "Load Z" when pad is not in circuit and "open circuit" when the pad is in the circuit.

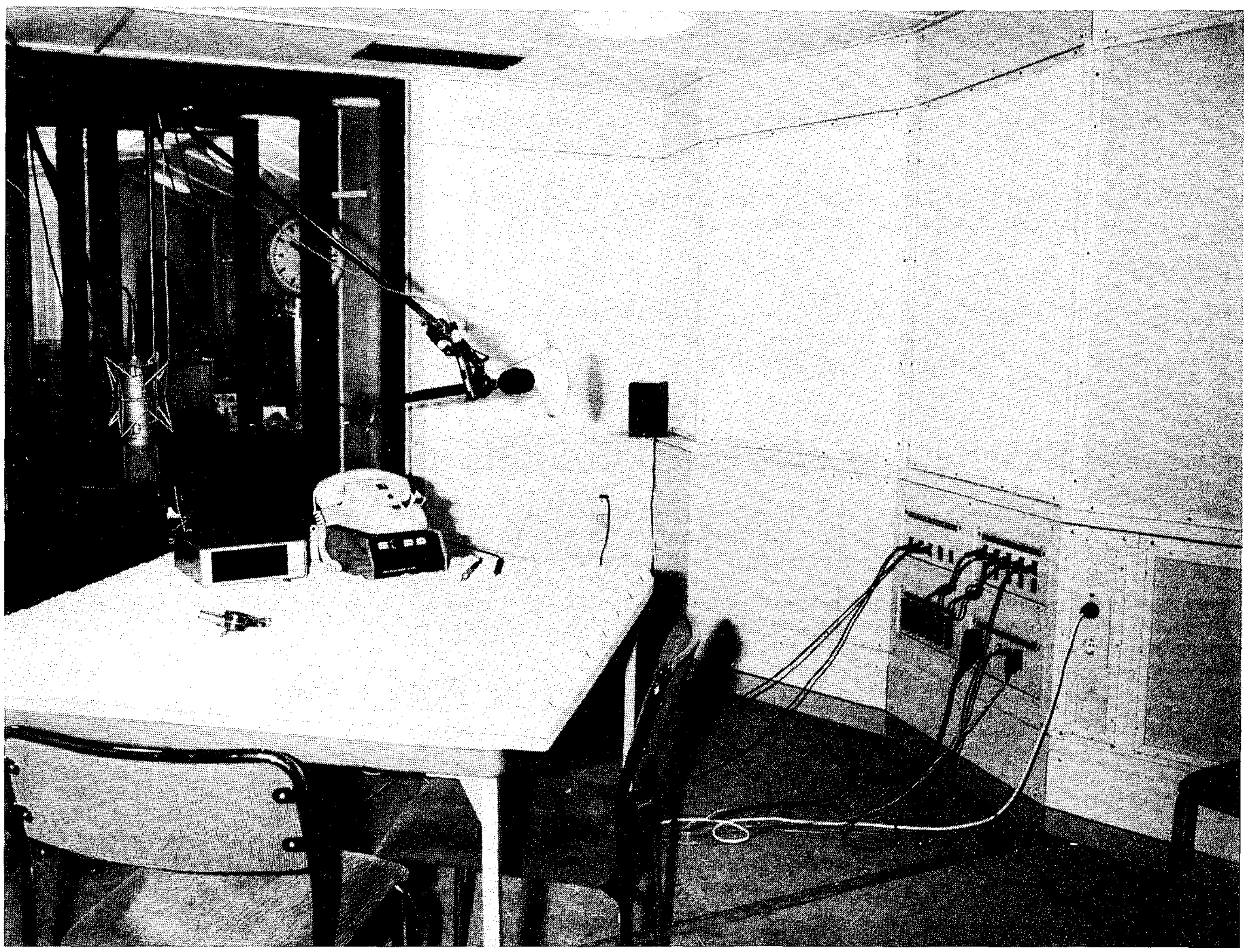
** Amplifier gain control is labeled in impedance values from 0.1Ω to 1.0MΩ. For a meter that indicates 0 dB for 0.775V across 600Ω the required gain for other Z values becomes: $\text{GAIN} = 10\text{ LOG}\left(\frac{600}{Z}\right)$

A TIME TO STUDY

Newsletter Volume 10, Number 3/4, page 30, is missing a heading. Please print in the heading: "A TIME TO STUDY," about two thirds down. The article starts, "Those of you already........"

SWEDISH RADIO COMPANY

Here's evidence of a careful observer and a clever applier of new ideas. Note the Sonex on the table top (no harmful reflections into microphones from there). Note even closer and you'll see the "darning hoop/ panty hose" wind screen in front of one of the microphones. Swedish Radio Company is not fooling around and neither is Lennart Nilsson. We are always pleased to hear of such applications from Syn-Aud-Con graduates. Mr. Nilsson and several of his colleagues have attended many Syn-Aud-Con seminars and workshops.



USING LOGARITHMIC SCALES DERIVED FROM LINEAR SCALES

FFT analyzers are based upon linear frequency analysis. They often, however, provide a logarithmic frequency scale display in addition to their normal linear frequency display. In generating such displays, they usually specify the frequency span displayed as so many decades (2.7 decades being a frequently encountered value). This span value means that the display goes 2.7 decades *below* the chosen high frequency (H.F.) limit.

The General Case Equation

$$\text{L.F. limit in HZ} = e^{(\ln \text{H.F.} - ((\text{N decades})(\ln 10)))}$$

For instance, if we choose 20 KHZ as our H.F. and 2.7 decades as our span, then the lowest frequency is:

$$\text{L.F.} = e^{(\ln 20{,}000)-((2.7)(\ln 10)))} = 39.91 \text{ Hz (i.e. 40 Hz)}$$

On the typical FFT with 400 lines of resolution, the resolution then becomes the linear frequency span divided by 400, i.e.: 20,000 Hz/400 = 50 Hz.

At an Fc of 100 Hz, this is a resolution that is twice as broad as the typical 1/3 octave analyzer (23% of the center frequency or in the case of Fc = 100 Hz, a bandwidth of 23 Hz). At an Fc of 1000 Hz, this same 50 Hz bandwidth is approximately 1/5th the 230 Hz bandwidth of the 1/3 octave analyzer.

How Many Octaves to a Decade

Since: $\frac{\text{H.F.}}{\text{L.F.}} = 10^{\text{N decades}}$ and: $\frac{\ln\left(\frac{\text{H.F.}}{\text{L.F.}}\right)}{\ln 2} = \text{N octaves}$

we can substitute and write:

$$\text{N octaves} = \frac{\ln\left(10^{(\text{N decades})}\right)}{\ln 2}$$

Thus 1.0 decade expressed in octaves is: $\text{N octaves} = \frac{\ln (10^1)}{\ln 2} = 3.32$ octaves

and 2.7 decades would be: $\text{N octaves} = \frac{\ln (10^{(2.7)})}{\ln 2} = 8.97$ octaves

Conversely, the number of decades in an octave becomes: $\frac{\ln 2}{\ln 10} = \text{N decades} = 0.301$ (i.e., the $\log_{10}$ of 2.0 is the fraction of a decade that represents a 2/1 ratio).

Summary

Choosing useful frequency resolutions can be estimated from previous knowledge of conventional constant percentage bandwidth analyzers (usually a higher resolution is not only possible but desireable with the new TEF® analyzers). Presentation of the data can be processed as log, linear, octave, decade, or variations thereof.

SOUND SYSTEM DESIGN SLIDE RULE

Our Sound System Design slide rule is an extremely valuable tool. We wanted to be sure everyone was able to use its full potential so we rewrote the instruction sheet. We hope you like it. We wrote it but Jan Kreitz made sure it was easy to read by preparing all the graphics.

Have we missed any types of problems that can be worked on the slide rule? Let us know. We can make up inserts for the instructions and include in future mailings.

CREDIT FOR GLEN BALLOU

One thing that pains us deeply is when we fail to give credit for an idea used in our Newsletters and Tech Topics. *Synergetic* Audio Concepts could not exist without an abundant sharing of ideas. And nothing would shut down this sharing of ideas faster than any hint of a "rip off."

While we gave credit to Glen Ballou in our yellow book and in Newsletter Volume 5, Number 1, we failed to credit him in Tech Topic Volume 10, Number 7, "Using Your Real Time Analyzer to Measure the Electrical Impedance." Glen designed, drew and wrote the instructions for the impedance interface box on page 2 (1/2 page).

MISUNDERSTANDINGS RE TRANSFER FUNCTIONS

Misunderstandings regarding transfer functions are legion. We can't cover all of them but we'd like to comment on just a few. The total transfer function consists of three component parts.

$F(f) = Fa(f) \cdot Fm(f) \cdot Fd(f)$ Where: $F(f)$ is the total T.F.

$Fa(f)$ is the amplitude component

$Fm(f)$ is the minimum phase component

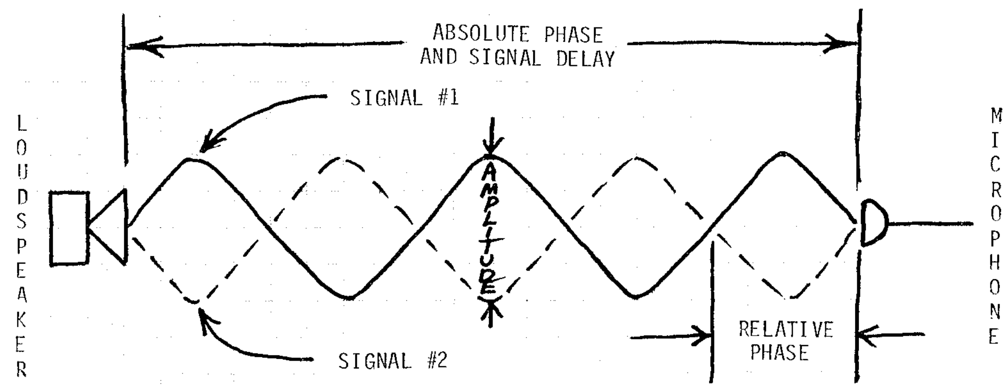
$Fd(f)$ is the pure delay component (an all-pass component)

Quite often the concept of "group delay" is mistakenly used as if it were $Fd(f)$. Heyser has clearly pointed out the inadmissability of this on the grounds of non-causal behavior of the group delay algorithm. (Exit Blauert and Laws Criteria when it is applied to loudspeakers.)

Syn-Aud-Con has wondered why $Fd(f)$ values of 10 to 20 usecs are so audible (using the new Sunn prototype micro second delay) in spite of what all the literature has to say on the subject. Mahlon Burkhard of Industrial Research Products, Inc., supplied the surprisingly simple explanation. The psychoacousticians use "clicks" (and often headphones) to study the effect of time delays. Syn-Aud-Con uses speech and music. The "clicks" never produce energy over enough time span to allow the energy from one "click" to "interact" with the energy of the other "click."

Speech and music is of sufficient duration that the time delay produces a variance in the amplitude response, $Fa(f)$. Since we don't listen to "clicks" as program material (or at least we hope you don't), their use in tests and the results obtained thereby are isolated academic anomalies having literally no application to sound system design, alignment, or perceived quality. The argument that we don't hear time differences because they manifest themselves as amplitude variations is an exercise in semantics. (Exit Blauert and Laws Criteria as a guide to time alignment.)

Phase and Polarity Illustrated



(A) Signals #1 and #2 have the same absolute phase.

(B) Signals #1 and #2 have the same relative phase.

(C) Signals #1 and #2 have the same signal delay.

(D) Signals #1 and #2 have the same amplitude

(E) Signals #1 and #2 *do not have the same polarity*.

Heyser has pointed out a startling fact when he showed how the variations in a minimum phase system can be fully modeled by a series of constant level all-pass networks ($Fd(f)$) which are all non-minimum phase. Indeed, this is precisely the behavior of sound through the air delayed by the velocity of the medium.

Finally, to go from the sublime to the ridiculous, two wavefronts out of a loudspeaker can have the identical absolute phase, the identical relative phase, the identical signal delay, and the identical amplitude and clearly be totally different. How? One is of the opposite polarity.

FREE NEWSLETTER SUBSCRIPTION

WOULD YOU LIKE A FREE NEWSLETTER SUBSCRIPTION FOR ONE YEAR? Send us original material for the Newsletter or a Tech Topic, no less than two pages of typed and illustrated material, and ideally four pages.

Tech Topic Volume 10, Number 1, "Termination, Impedance Matching and the Maximum Power Transfer Theorem in Audio Systems," by Jim Brown is an example of excellent material -- well thought out and expressed.

Have you designed and installed a communications system that you have a good glossy picture of supported by system design calculations? It is a good idea to send a very short description of what you propose before going into full production.

XIT GROUNDING ROD

Harold Lindsay of Ampex fame enjoyed being with us for each of our classes in San Francisco until his death in 1982. He told us about the XIT Rod and said he specified it in his audio and acoustical practice.

We are reproducing an article here from *Design News* which we feel is important. If you don't know about XIT, their address is P. O. Box 128, Beaumont, CA 92223. (714)845-3986.

Design IDEAS

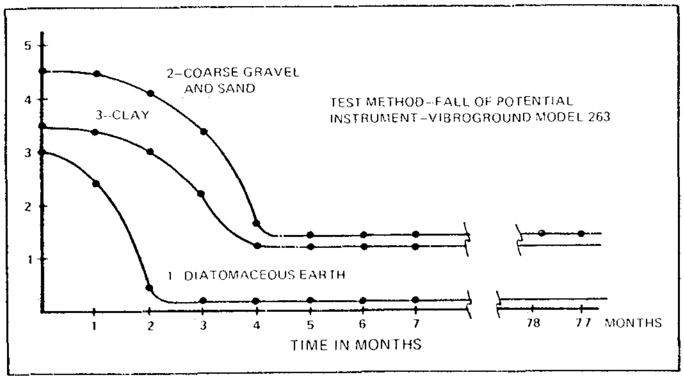
Reprinted from DESIGN NEWS February 7, 1977

'Electrolytic roots' improve electrical grounding

Chemically charged grounding rod uses moisture from air, grows "electrolytic roots" that rapidly decrease grounding rod-to-earth resistance

E. J. Stefanides, Central States Editor

A new type of electrical grounding rod is designed to use the moisture in the air as a means for providing rapid but permanent improvements in conductivity of the electrical interface between its surface area and the surrounding earth. In use, air is pumped by atmospheric pressure changes into the hollow tubular body of the rod through holes near the top. Moisture in this air is extracted by condensation and trickled through a bed of coarse granulated metallic salts. In process, a very small quantity of the salt is dissolved to form an electrolytic solution that is accumulated in a reservoir at the bottom of the rod's tubular body. As time passes the excess electrolytic solution is pumped out through bleed holes near the bottom of the tubular body by the same atmospheric pressure change phenomenon. This excess solution, seeping into the earth at a rate of 7 or 8 drops per day, establishes "electrolytic roots" that provide a very large reduction in the electrical resistance between the rod and the surrounding earth.



Pumping action (below) provided by changes in atmospheric pressure is indicated by arrows. High humidity conditions are not necessary, though extra long rods (i.e. 40 ft or more) may be required in arid regions.

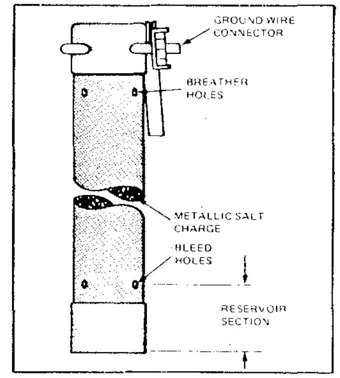
Final level of resistivity and time required to reach this steady state level will depend on type of soil and method of installation. Tests conducted for 6-1/2 years show no change in straightline values of resistance.

The nature of this "electrolytic rooting" action is such that within a relatively short time (2 to 4 months) the resistance will drop to a small fraction of its low (3 to 5 ohm) initial value.

Further, since the rate of seepage and the rate of salt dissolution are both very low and the corrosion resistance of the rod's tubular jacket is very high, it has been estimated that the rod should be capable of maintaining the low-resistance electrical interface across a service life of at least half a century.

These grounding rods are design developments of XIT Rod Co., Covina, CA, patented in the U.S. and U.K., and with patents pending in other countries. The rods are manufactured under the registered trademark "XIT Rod" and marketed within the United States by various distributors, including Designed Mining Products Corp., Belle Vernon, PA (East Coast distributor). They have been accepted for UL listing and meet the requirements of the Revised National Electrical Code.

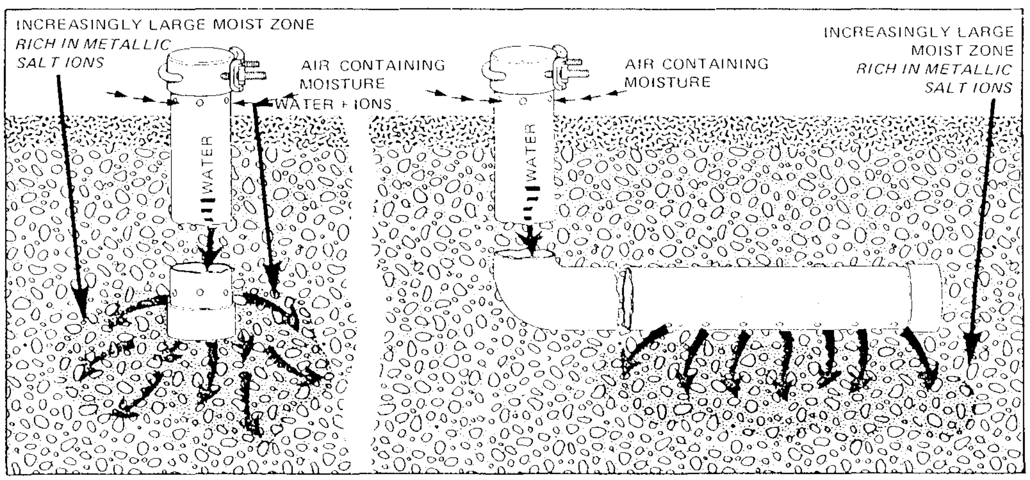
The rods are manufactured of type K copper tubing in lengths from 8 to 20 ft (2.44 to 6.20m), then filled with the metallic salt. Slip-on caps are subsequently brazed in place and provision for electrical connection is made by installing a U-bolt with pressure plate and prevailing torque nuts at one end of each tube. Four holes are then drilled about 2 inches (5.1 cm) from the bottom of the tube to allow seepage of the electrolytic solution.



Metallic salt charge is added by manufacturer before shipping. Tape is applied over breather and bleeder holes to prevent loss of salts during shipment.

Rods to be installed in areas that have thin layers of soil over bedrock are designed to be buried horizontally in trenches and are bent to an L-shape. The 20 to 30 ft (6.20 to 9.15 m) horizontal leg of this L-shape is cross drilled at intervals along the length to provide for the required "electrolytic rooting" action.

During installation of the rods a salt solution is used to saturate the earth around the tube. This saturation establishes an initial electrolytic bond between the rod and surrounding earth and provides a starter solution for the long-term "electrolytic rooting" action.



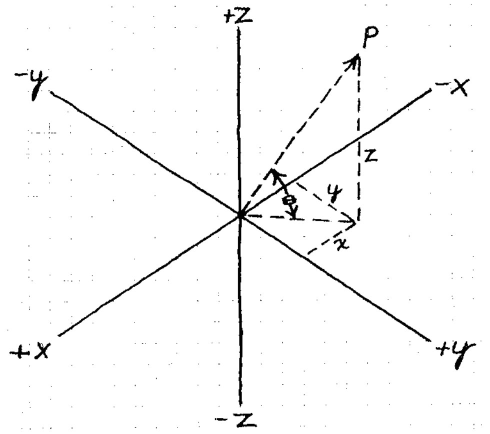
Grounding rod is equipped with U-bolt pressure plate connection designed to accommodate #8 to #3/0 AWG grounding wires. All but top 4 inches (10.2 cm) of rod are buried in ground.

MORE USEFUL MATH TOOLS

USEFUL FORMULAE USING "e"

$$e^x = \cosh x + \sinh x$$
$$e^{-x} = \cosh x - \sinh x$$
$$\sinh = \frac{e^x - e^{-x}}{2}$$
$$\sinh^{-1} x = \ln\left(x + (x^2+1)^{1/2}\right)$$
$$\cosh x = \frac{e^x + e^{-x}}{2}$$
$$\cosh^{-1} x = \ln\left(x + (x^2-1)^{1/2}\right)$$
$$\tanh x = \frac{e^x - e^{-x}}{e^x + e^{-x}}$$
$$\tanh^{-1} x = \frac{1}{2} \ln\left(\frac{1+x}{1-x}\right)$$
$$e^{ix} = \cos x + i \sin x$$
$$e^{-ix} = \cos x - i \sin x$$
$$\cos ix = \cosh x$$
$$\sin ix = -i \sinh x$$
$$\cosh ix = \cos x$$
$$\sinh ix = -i \sin x$$
$$1 = \sin^2 x + \cos^2 x = \cosh^2 x - \sinh^2 x$$

VECTOR NOTATION



$$P = \sqrt{x^2+y^2+z^2} \qquad \text{((P) IS THE VECTOR LENGTH)}$$
$$\theta = \tan^{-1}\left[\frac{z}{\sqrt{x^2+y^2}}\right] \qquad \text{((θ) IS THE VECTOR ANGLE)}$$

BESSEL FUNCTIONS

$$J_0(x) = 1 \frac{x^2}{2^2} + \frac{x^4}{2^2 4^2} + \frac{x^6}{2^2 4^2 6^2} + \cdots$$
$$J_1(2x) = x\left[1 - \frac{x^2}{1.2} + \frac{x^4}{1.2.2.3} - \frac{x^6}{1.2.3.2.3.4} + \cdots\right]$$
$$J_n(x) \; \frac{x^n}{2^n . n!}\left[1 - \frac{x^2}{2(2n+2)} + \frac{x^4}{2.4(2n+2)(2n+4)} + \cdots\right)$$

HOW BIG A TIME WINDOW? HOW LOW A FREQUENCY?

The time-bandwidth product used in conjunction with FFT analyzers reflects a "time window" philosophy that can be applied to any measurement requiring a high degree of frequency resolution.

$$T_R \cdot F_R = 1.0 \text{ (Time-Bandwidth product)}$$
$$T_R = \frac{1}{F_R} \quad \text{(Time window)}$$

If we require a high degree of F_R at 30 Hz then the time window should be on the order of $\frac{1}{30}$ = 0.03 secs.

One wavelength at 30 Hz is: $\lambda = \frac{1130}{30} = 37.67'$

and the time taken to develop one full wavelength is: $\frac{37.67'}{1130'/\text{sec}} = 0.03$ sec.

Therefore, what the time-bandwidth product is telling us is that we need at least one wavelength of a given frequency for a high degree of frequency resolution.

The ETC measurement tells us at a glance the distance and the time between the direct sound and the first reflection, thus enabling us to calculate the lowest frequency we'll be able to measure with high resolution.

$$f_L = \frac{1130}{d} \quad \text{or} \quad f_L = \frac{1}{t}$$

"CONTROLLED TRAVEL PATH DESIGN"

Three stages in the development of a new idea:

1. It won't work.
2. I knew it all the time.
3. I developed it.

Quoting from the August 1983 issue of *R-e/p*, "Westlake Studio C - Applications of Controlled Travel Path Design":

"Westlake Audio president Glenn Phoenix would be the first to acknowledge that his innovative design for the new Studio C at the facility's Santa Monica Boulevard, Los Angeles, location is based upon a distillation of the main acoustics theories and practical construction techniques developed over the last decade or so. But with some interesting and revolutionary advances and fundamental improvements.

"While the recording area purposely features a reasonably live acoustic treatment -- a trend that is appearing in an increasing number of studios these days -- it is in the control room design and implementation that the fundamental differences between the new Westlake and "traditional" designs lie.

"'The philosophy behind our new 'Controlled Travel Path,' or CTP design,' Phoenix says, 'is that the angles of reflection and surface acoustics treatment are arranged to restrict early sound reflections into the listening or monitoring environment. The result is a broad, high-definition soundfield across the central area of the control room, with a RT_{60} that is consistent at about 0.3 seconds within the low- and mid-frequency range, falling to around 0.25 seconds above 8 kHz.'

"The underlying key to the new design approach, Phoenix offers, is to minimize the repetitive paths for sound reflection around the surfaces of the control room, including both standing waves between parallel walls, floor, ceiling, and other areas, plus the odd reflection and refraction modes between room corners, etc. By controlling the ways in which sound waves from the monitor loudspeakers and secondary sources can repeatedly 'bounce' or be reflected and refracted around the room, a uniform, well-behaved environment possessing a diffuse soundfield with a consistent decay characteristic can be created.

"And if there appears at first sight to be a close kinship with at least the fundamental approach of Chips Davis and Syn-Aud-Con's Live End/Dead-End™ design philosophy -- in which, in essense, the console monitoring position defines a line across the control room in front of which the acoustic treatment is purposely highly absorbent, while behind it is configured to be reflective -- Phoenix would readily agree. But with certain important reservations, however."

Of course! We're in the third stage of development of a new idea.

In the same article a new *five*-way Westlake monitor is discussed. It has crossovers at 200 Hz, 800 Hz, 3.2 kHz and 10 kHz. Quoting:

"Housed in a 19-inch rackmount case, the high-resolution crossover unit contains two channels of four-way, *24 dB per octave* (italics mine) filters, which minimize driver-to-driver interferences, and optimize bandwidth control.

"As Glenn Phoenix explains, '*The 24 dB per octave slopes* (italics mine) are used for the crossover because you need to get the selected signal band into -- and out of -- a speaker driver quickly, while maintaining complementary phase, both above, at, and below the crossover points. This allows for a virtually seamless transition of sound from one driver to the next.'" (Editor's Note: The "time smear" thus introduced has to be noteworthy.)

TDS-ETC measurements brought about LEDE™ and CTP designs. It won't be long before there are almost 100 TEF® (a Crown/Tecron trademark) analyzers in the public's hands. It won't take long for TEF® to revolutionize monitor loudspeaker design.

dB ABUSED AGAIN

Capp Loughboro, Ventura, California, sent a copy of an article that appeared in a CATJ magazine in July 1982 written by a Wavetek engineer on the myths surrounding the decibel. The author misinformed the maximum number of readers possible. Intitled the "Gravelization of Spinach" (if you're going to be wrong, be "cute" and be "aggressive"). The writer of this article must have been raised by permissive parents as he is firmly convinced that if enough people use an incorrect notation *then the rules should be changed*.

His most heinous offense, however, is in carefully illustrating audio amplifier gain as the difference between the power in dBm dissipated at the input to the power in dBm dissipated at the output (i.e., the power gain) rather than explaining the concept of "available input power" and correctly calculating the "insertion gain" of the device. Equally misinformed readers wrote in correcting his numerical errors and agreeing to the really gross errors.

As usual, the editors stood to one side in totally blissful ignorance convinced that publishing these experts was "training" their readers. We'll never know the correct gain of the example cited because he never gave the source impedance.

WHO WROTE IT?

The Syn-Aud-Con Newsletter is written by Don and Carolyn -- unless the article is signed. The following story will give you the clue as to whether Don or Carolyn wrote the article.

We started a hi-fi shop in Lafayette, Indiana, in 1951 called The Golden Ear. (The first year of business we sold one hi-fi system and *could* have made a good living selling seed corn and hearing aids.)

We loved our hi-fi shop and spent a lot of time in it listening to music. (To our knowledge, for many years we were the only hi-fi shop in the Middle West outside of Chicago.) One night a customer was listening to a new Cook record of Carlos Montoya. Our friend, who had worked in Mexico, said he could visualize the bull-fights and all the color associated with them as he listened to the music; Don said he could see the court-yards of the hacienda and smell the flavor of Mexican cooking; Carolyn said she saw a man sitting on a bare stage playing a guitar.

So if the article is "purple" with color, Don wrote it; if it is "bare" of imagery, Carolyn wrote it.

SMILE

HE didn't call them ten suggestions!

SERIOUS

Failure is the path of least *persistence*.

CLASSIFIED

FOR SALE: Yamaha Power Amps. Four P-2100 stereo 120w/cl amplifiers; two bi-amp 270 1/3 octave equalizers. Asking one half of retail or???? Trade??

Call Jeff Loether (800)638-6707, Ext. 7159 (9am - 4pm EST) or (301)762-4413 (message service) anytime.

WANTED: HP tape drive.

Call Jeff Loether (800)638-6707, Ext. 7159 (9am - 4pm EST) or (301)762-4413 (message service) anytime.

FOR SALE: 3 each 531A Tektronics oscilloscopes with CA plug in (dual trace) - working. . . . . $400.00 each

1 each Panoramic Sonic Analyzer LP1A with companion sweep generator G2A $250.00

Contact Brock Jabara, Galaxy Audio, 625 E. Pawnee Avenue, Wichita, KS 67211. Phone (316)263-2852

FOR SALE: New, never used 1970 Altec - HP 8050A real time analyzer.

Contact Phil Langdon, Audio Resources, 778 Burr Oak Drive, Westmont, IL 60559. Phone: (312)655-1180

EMPLOYMENT OPPORTUNITY: We have heard of the need from a manufacturer for a Chief Engineer and an Electronics Engineer. Also, a sound contractor needs a Designer. If you are interested, send a letter to us (*don't* call) and we'll forward.

Syn-Aud-Con, P. O. Box 669, San Juan Capistrano, CA 92693

SYN-AUD-CON SPONSORS

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

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