SYNERGETIC AUDIO CONCEPTS

SYN AUD CON

newsletter

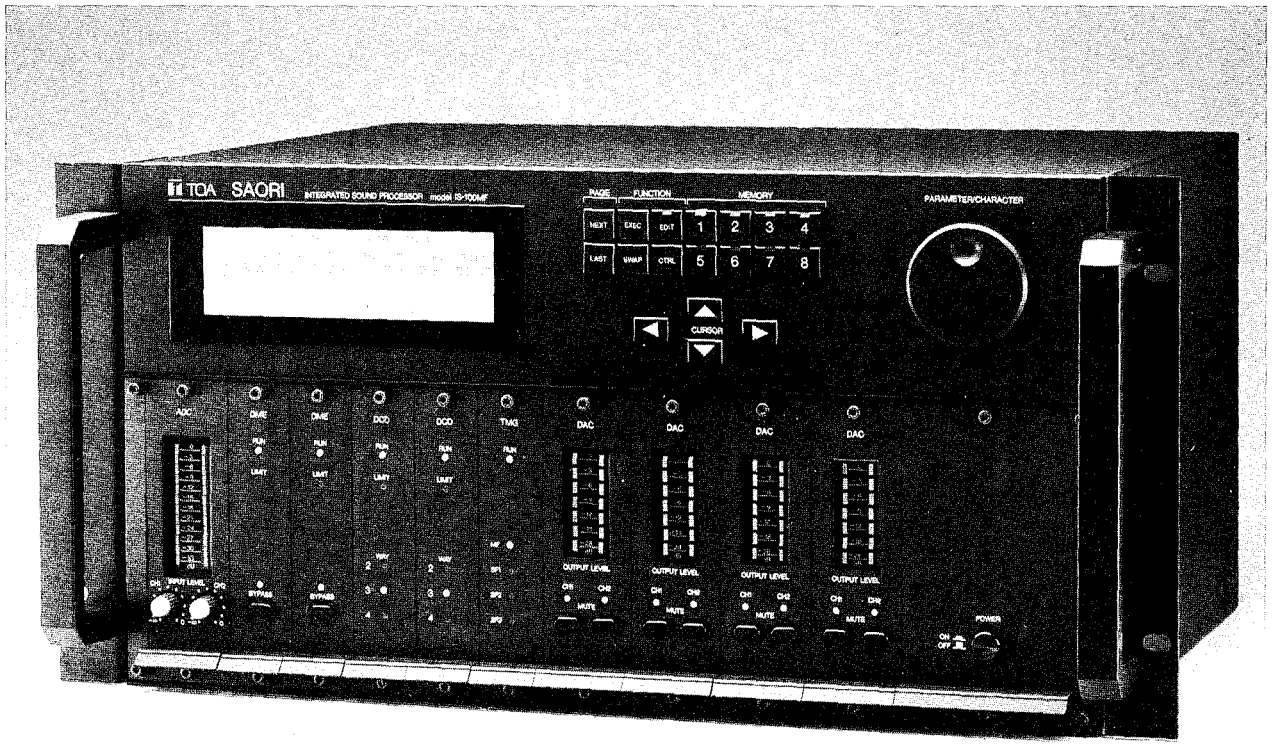
Volume 17, Number 1
Fall, 1989

TOA

SAORI:

Digital Signal Processing for Audio Systems

The Digital Promise Delivered



SYNERGETIC
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AUDIO CONCEPTS

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.

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Carolyn Davis

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The information conveyed in this NEWSLETTER has been carefully reviewed and believed to be accurate and reliable; however, no responsibility is assumed for inaccuracies in calculations or statements.

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TABLE OF CONTENTS

SAORI – Digital Signal Processor by TOA

In 1973 we predicted with a great deal of confidence that within a decade the world would go digital. Well, the world's going to go digital all right, but it took two decades instead of one.

Syn-Aud-Con owns a DAT recorder (the Sony TCD-D10 PRO model) and we have just seen and measured what will go between its output and the input of our power amplifiers. It's a digital sound processor from TOA called SAORI.

SAORI is a truly spectacular two channel, fully digital processor that includes a 30 band 1/3-octave graphic equalizer with selectable filter Q's of 3.5, 5.0, and 7.0; four narrow-band rejection filters with Q's selectable between 30 and 70. All four filters set on one frequency provides 72 dB of attenuation.

A fully digital crossover system is included that allows 2-way, 3-way, and 4-way modes and provides the choice of Butterworth, Linkwiltz-Riley and Bessel configurations with slope rates of 12, 18, and 24 dB/octave. There is also a "horn equalizer" for compensating the high frequency rolloff of CD horns with the inverse of their slopes.

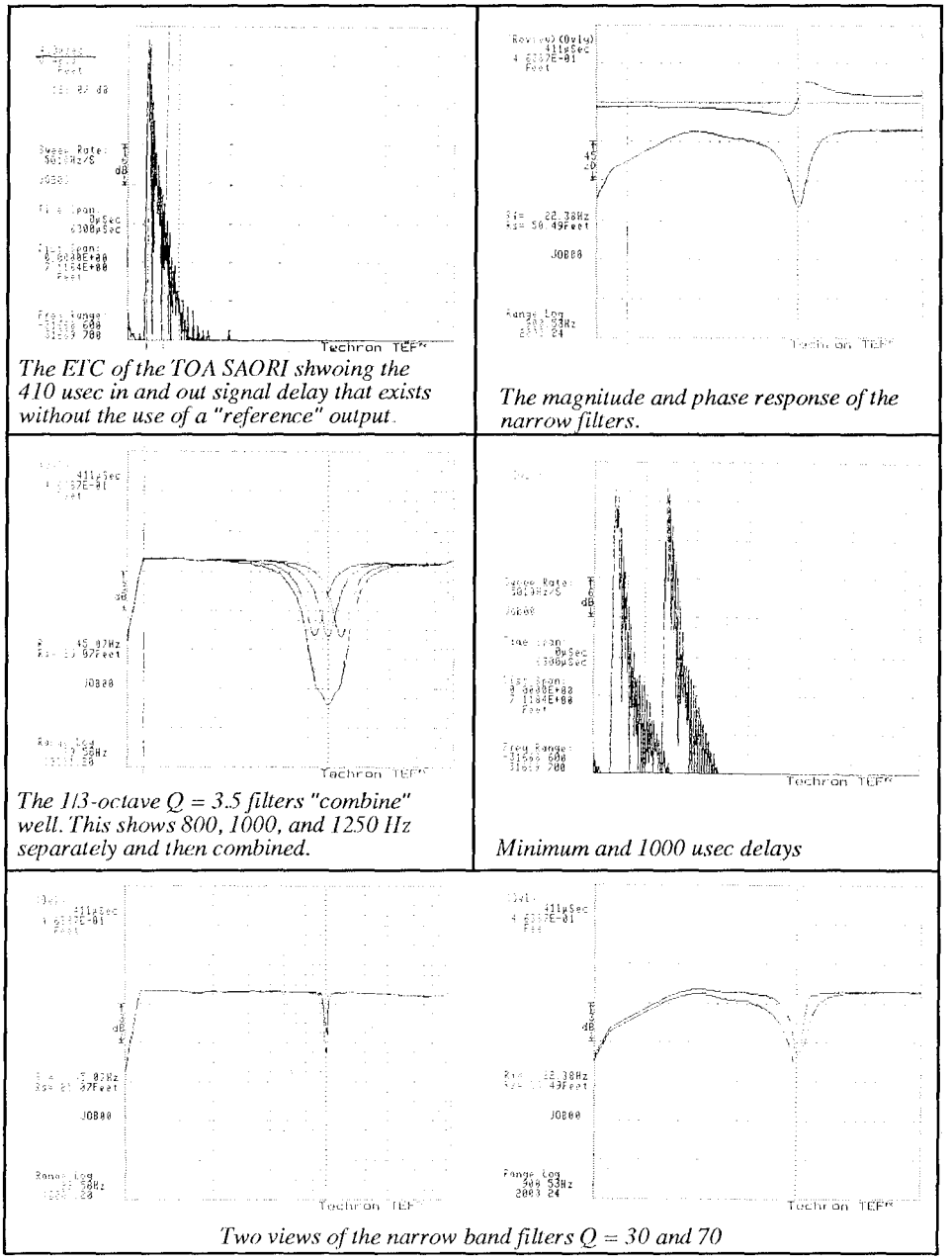
Very accurate signal delay is available in 21 usec steps. The signal delay is provided for each divided band for alignment purpose. The processor has two channel inputs and eight channel outputs.

We measured the input impedance as 10 KΩ (what's also marked on the input connector). The output connectors were marked 600Ω (but measured 60Ω). The input and outputs have voltage ratings of 1.23 volts and when connected to a load of 600Ω or greater, makes the unit a unity gain device.

We found that it takes 410 usecs to go from input to output with zero delay settings, thereby necessitating a "reference" output to feed the undelayed devices in the system.

A large easily read LCD panel makes operation quite simple. Our measurements reveal that the device has absolute accuracy in terms of any imaginable sound system usage. It is fascinating to realize that a digital time domain device can, by mimicking the complex impulse response of a series of analog devices, produce a frequency domain signal of the desired characteristics with even greater accuracy than the collection of individual analog devices.

SAORI marks, we believe, a new TOA stance in the United States. This unit, we are told, is the first in a series designed to meet the most sophisticated needs of professional audio contracting firms that compete for the largest and most complex jobs. It literally has no peer currently in the U.S. market. To miss getting one of these units as early as possible is to miss seeing the future now. This is the processor that will be to the TEF analyzer what the 1/3-octave filter set was to the real time analyzer.



The ETC of the TOA SAORI shwoing the 410 usec in and out signal delay that exists without the use of a "reference" output.

The magnitude and phase response of the narrow filters.

The 1/3-octave Q = 3.5 filters "combine" well. This shows 800, 1000, and 1250 Hz separately and then combined.

Minimum and 1000 usec delays

Two views of the narrow band filters Q = 30 and 70

"Cables and the Amp—Speaker Interface" by Dr. Greiner

"I have seen no scientific or statistically significant studies which show that all the different designs have any significant or audible effect whatever on the transmission of audio frequencies in these cables."

Eugene Pitts, editor of *Audio Magazine*, has lived up to the highest standards left by founding editor, McProud, when Mr. Pitts published R. A. Greiner's "Cables and the Amp-Speaker Interface" in the August 1989 issue of *Audio Magazine*.

Dr. Griener's work in audio is always of the highest quality and presented in a manner that only minds who enjoy sharing fully with others can achieve. No pompous nit picker here. Spade's a spade. Wit and skill shape the facts into plain entertaining English.

The Dr. Grieners, Dr. Patronises and others of their caliber and integrity are sufficiently rare in academia today to merit special mention and special attention.

A preface of this article includes the following:

"The substance of the paper, however is based on electromagnetic theory, and no amount of advertising in the past 10 years has changed that base."

What we see going on with esoteric cables was in 1889, patent medicine. What's interesting this time is that many medical doctors are the victims.

W. C. Fields said, "Never give a sucker an even break," and "Never smarten up a chump." That's good advice because humans love to be duped. Those of you that don't enjoy being had can take genuine pleasure in Greiner's article and use it to help pull down at least a portion of the Temple of Hype. There are genuine mysteries in the human experience. Cable effects in audio is not one of them.

The article is nine pages long and an important article to read and save. I would like to quote a few paragraphs from the article:

"If you determine the absolute worst-case frequency limits for a 10-meter-long run of cable, the effects will occur at frequencies well above audibility."

Further on,

"With the majority of cables, 100-Hz and 10-kHz signals do not actually arrive at the same time. However, their arrival times are only fractions of a micro-second apart."

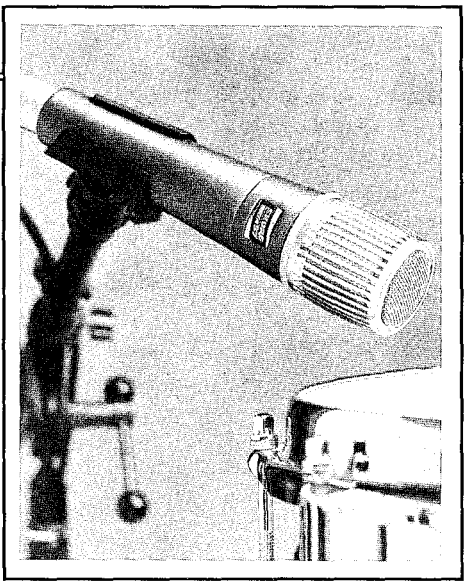
Dr. Griener suggests a few "common sense and sensible precautions" to be followed:

"Choose a cable that has reasonably low resistance - say, less than 5% of the lowest resistance of the loudspeaker at any frequency. Choose a twisted pair of wires to reduce or eliminate any possible crosstalk between wire pairs or from parallel power cords. Make connections on each end with proper spade lugs or screw terminals which can be firmly tightened...look for a mechanically sound connector. Too many connectors are mechanically unreliable, with possibly one of the worst being the common RCA phono connector used on consumer equipment. Professional equipment uses XLR or BNC connectors for good reason. Gold-plating is quite common today and certainly does no harm."

Shure's Beta 58 and Beta 57

Shure has developed two new microphones called the Beta 58 and the Beta 57. We have been using a Beta 58 in our farm classes, which include live equalization demos. We have found them to be superior both in voice quality and in their ability to reject early reflected energy before it gets too diffused. In our opinion a worthwhile improvement in every way.

We have seen a lot of microphone placement such as being used with the Beta 57 in the photograph. Watching the recording engineers go through their machinations we realize that they are "tuning" for the desired comb filter effect. In cases like this both the microphone's directivity, and its placement can be used interactively to obtain considerable coloration. That's why recording is still an art and a science.



Shure's Beta 57 Supercardioid Dynamic Microphone

Analog Room Acoustic Simulator

Barry McKinnon is the sales manager for DSE Production Equipment and Services, Ltd., Calgary, Alberta, Canada

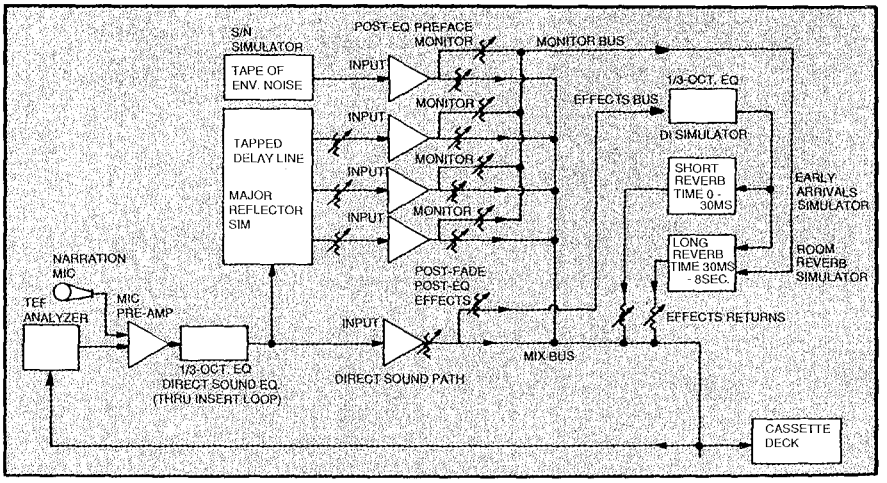


Figure 1. This block diagram is an example of the hardware required for a sound system/room simulation. The TEF feeds the mic input that will eventually be used to narrate the demonstration tape. The signal processing that will simulate the direct sound equalization and the major reflections are in the insert loop.

Barry McKinnon has detailed an analog room acoustic simulator in an excellent article in *Sound and Video Contractor*, June 20, 1989. He describes how he not only uses it to make simulated TEF measurements, but talks through it so the customer can hear a before and after acoustic treatment or sound system installation.

Having seen and heard similar systems done by VMA Peutz back in 1981, we know that they can be made to sound very realistic.

Barry's system controls the direct sound, the far field directivity index, the early decay and the late reverberation plus many other room-system parameters. Barry measures, for example, a real space (existing construction) and then duplicates the ETC in the simulator by using delay lines, and audio console, clever use of equalizers, and artificial reverberation devices.

The diagram in Figure 1 helps clarify Barry's approach. The ETCs in the original article of the simulations compared to the actual room measurements are impressive. We are reproducing Figures 7 and 8 with a quote from Barry's article:

> *"Comparing the two you can see the higher direct arrival levels of approximately 6dB, and a noticeable difference in the early arrival density and measured reverb energy and character. This is text book-type material showing the benefit of high Q horns. High-Q horns reduce early arrivals and they reduce excitation of reverberant space."*

We recommend the article be read in its entirety.

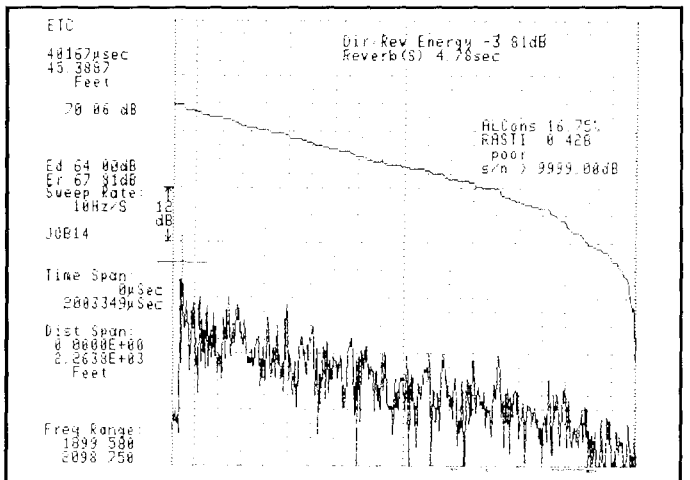


Figure 7. Horn/loudspeaker measurements taken at an ice rink using a 90° x 40° horn as the sound source. The horns were placed in the penalty box, pointing toward center ice. The microphone is located at center ice.

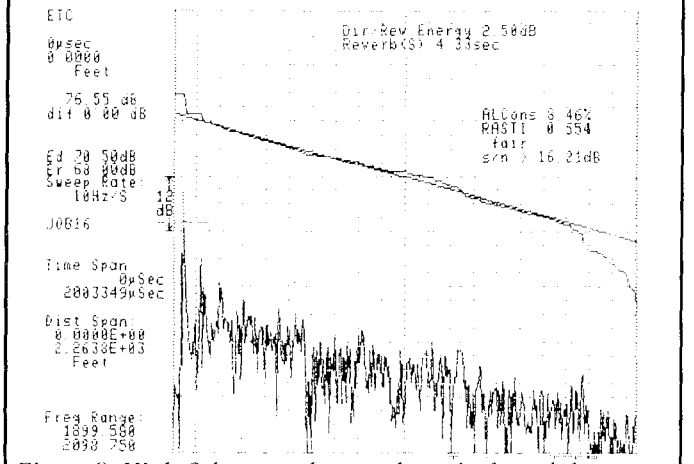


Figure 8. High-Q horns reduce early arrivals and they reduce excitation of reverberant space, as shown by the 40° x 20° horn.

Grounding & Shielding: System Adjustment Procedures for Sound Rinforcement

by
Jim Yerges, Ph.D., P.E.
Yerges Acoustics, Downers Grove, IL

Editor's Note: We have a lot of respect for Jim Yerges. We asked him to share his thoughts on grounding, shielding and system adjustment. We think you will find the following as interesting and useful as we did.

I do not claim any unique expertise in the area of grounding and shielding. The two accompanying figures pretty much summarize our standard approach, which has kept us out of trouble. It represents the evolution of a method Bob Ancha showed us years ago, and it seems to be reasonably consistent with what we see in trade journals.

Our theory about equipment selection and impedance matching is pretty simple. We try to use only pro-audio pieces with balanced outputs looking at pro-audio pieces with balanced inputs. The actual output impedances are typically very low—often less than 200 ohms. The actual input impedances are typically very high - usually in excess of 18,000 ohms. We have recently begun to allow active balanced inputs in lieu of transformers as long as everything is in the same rack, and have not had trouble yet. When confronted with an unbalanced input or output, such as a not-quite pro-audio tape deck as a result of a not-quite pro-audio budget, we use a fairly reliable little scheme for interconnecting to balanced components. (We expropriated the detail from a University Sound newsletter.)

We do not use patch bays unless they are necessary, because they are prone to intermittent open circuits as they become dirty. We are watching the new routing switchers as they reach the market, but have not yet used them on a large scale.

We absolutely insist that signal level cables be routed from each device to a clearly labeled terminal strip, and only then to the next device. This tidies up the wiring and creates probe locations for system setup.

System Alignment

When we adjust a system, the first order of business is time alignment, using the TEF machine. We do this before shelving, because the output of the low-frequency loudspeakers must be greatly increased to compensate for the limited energy in their sweep band, which is much narrower than the sweep band of the mid-frequency or high-frequency devices. (Apologies to Ed long for the usurpation of Time-Align, but "signal synchronization" just doesn't roll off the tongue. I suspect that time alignment will become to Time-Align as fiberglass is to Fibreglas.)

System Equalization

Only after time alignment, we shelve and equalize. We have not had much success in our attempts to equalize to the direct sound using the TEF machine. First, it is maddeningly difficult, because the TEF is a constant bandwidth analyzer. Second, the resulting equalization has too much bass left in it to sound natural. Apparently, listeners hear and integrate some reverberant sound below approximately 500 Hz. In well-behaved rooms, such as auditoria with upholstered seats, we have had better luck with the following scheme:

First, find the critical distance from the loudspeaker cluster in the frequency range centered around 500 Hz. This is pretty straightforward using the Schroeder integration and the cursor on the TEF machine.

Second, determine how high above the seats the microphone must be placed to avoid a reflection which is strong in comparison to the direct sound. Minus 10 dB would be nice, but not necessarily attainable. In order to perform this measurement, we have had to modify an aluminum extension pole for use as a microphone stand.

"A good acoustical consultant is somebody who is old enough to have already made—or at least already seen—every imaginable mistake, but clever enough not to repeat one. The consultant comes to the client with forty years of scar tissue."

Third, place the equalization microphone on the extension pole. Continuously traverse the microphone across the imaginary surface located at the approximate critical distance from the cluster. Let the real-time analyzer

sample an ensemble of at least 256 spectra. With our machine—A Rion time-compression unit—this process takes approximately 2 minutes. However, the measurement is extremely repeatable, so the equalization process moves forward in an orderly fashion.

When all of the signal level devices are adjusted so that the tops of their individual dynamic ranges are aligned, we figure they are as far above their cumulative internal noise levels as is feasible.

Fourth, cautiously flatten the middle portion of the curve. The loudspeakers will determine the frequencies at which the curve begins to fall off. Do not attempt to boost the high or low-frequency extremes of the response by more than a few dB, because you will not like the sound. We used to insist on cut-only equalizers, but some of the newer cut-boost equalizers are easier to use and sound fine as long as you do not try for more than approximately plus-or-minus 3 dB boost or cut.

This little methodology evolved as a result of our definition of a "successful" equalization. We ask a reader to stand on the stage or chancel platform and read in a normal voice, with the reinforcement microphone tucked behind his back. We stand approximately four feet in front of the reader, and attempt to memorize the sound of his voice. Then we quickly retreat into the auditorium as the talker raises the microphone to within approximately 12 inches if his lips. If the reinforcement system sounds just like the unreinforced reader did at close range, we feel that we did a good job.

We have never enjoyed noteworthy success in our attempts to significantly improve gain-before-feedback using third-octave equalizers. Similarly, we have been underwhelmed by the usefulness of narrow-band equalizers for feedback control, except when the microphone location was absolutely fixed and permanent. Careful loudspeaker selection and location and control of room acoustics have been more productive. Time alignment (or the lack thereof) seems to explain some memorable fiascoes.

Music Systems

Music systems are easier. We ask the owner, "Do you want more thump? Do you want more sizzle?" Then we give him what he wants. (Perhaps I am not a man of principle.)

Level Setting

After equalization, we set levels to push the noise floor of the equipment as far below the operating level as possible. First, turn all of the system gains (except the amplifiers, which are already shelved) all the way down. Put a 1 millivolt sine wave into the mixer input gain, and bring the mixer input and output gains up together, until the output of the mixer, viewed on a scope, begins to show clipping. Back off the gains until the clipping just disappears. Move on to the next device and repeat, up to but not including the limiters. (This example assumes that the limiters come immediately before the power amplifiers. Do not whack the power amplifier inputs with a clipped input unless the loudspeakers are disconnected!) When all of the signal level devices are adjusted so that the tops of their individual dynamic ranges are aligned, we figure they are as far above their cumulative internal noise levels as is feasible.

A word is in order about the equalizer. Try to pick a test frequency in the minimum cut (or maximum boost) pass band of the equalizer.

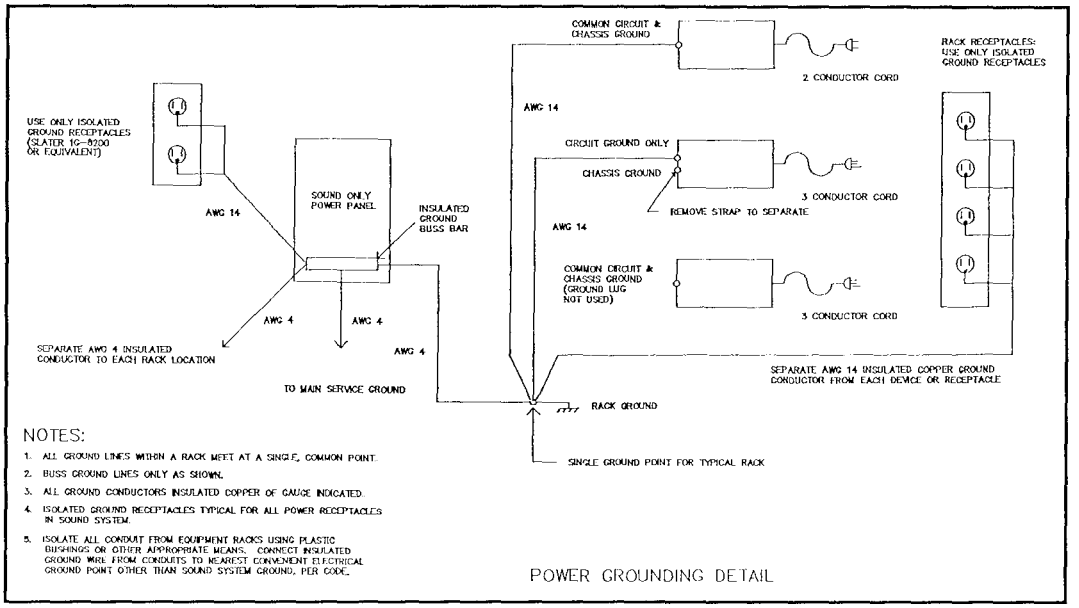
Limiters

Now we come to the limiter(s). We make a philosophical decision about what we are protecting against, (it depends on the application), and select an attack mode and threshold.

The same basic methodology applies to simple systems and to three-way and four-way systems.

Harmonic Distortion

We do not typically test a system for harmonic distortion unless something is wrong and it sounds bad, or there is a dispute with the contractor. Then we do a standard test using a nulling analyzer with resistors substi-

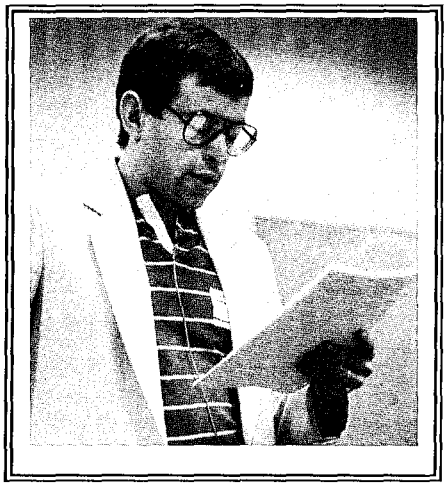


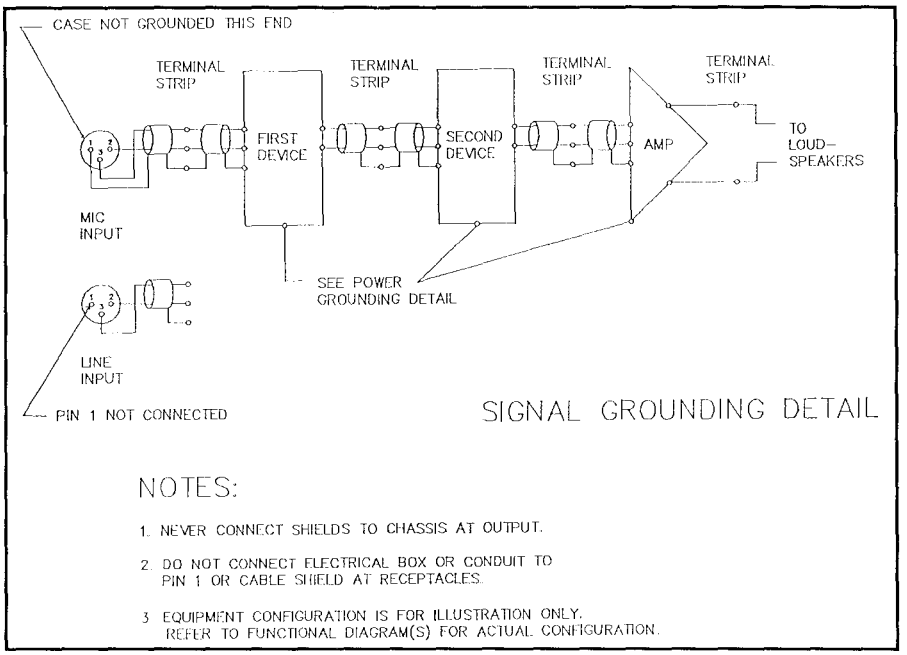
POWER GROUNDING DETAIL

tuted for the loudspeakers. Similarly, we do not test for "system noise floor" or "residual noise" unless hiss or hum are audible and/or there is a dispute over performance. Again, it is a pretty standard test, with a shielded, shorted mixer input. (Maybe this short should actually be a shielded 150 ohm resistor . . . I haven't really thought about it.) The "residual" voltage envelope at the output of the power amp, measured with a scope, must not exceed some allowable fraction of the full power output voltage. This fraction is usually expressed in decibels.

RF and Oscillation

Finally, we turn on all of the theatrical lighting, the wireless microphone transmitters, the hearing impaired system transmitter, and anything else which looks potentially troublesome. We then poke around with a 5 MHz oscilloscope and the probes the scope manufacturer recommended, and hope that we do not find any oscillation or RF pick-up. We have been very lucky about this when the systems are correctly grounded and shielded. However, we have seen some disasters where grounding has been inadequate. For example, a cathedral in Chicago periodically gets brief audio visits from a local AM "shock radio" deejay.





Summary

These remarks constitute an oversimplification. There are situations and equipment pieces which must be treated as special situations. Moreover, these remarks are not offered as a definitive statement of how everybody ought to ground and adjust sound systems. Rather, they are a description of things that have worked, so far. I could write an equally long list of ideas, procedures, systems and products which, at one time or another I enthusiastically embraced, but had to drop like a hot brick after they burned me.

My father used to claim, "A good acoustical consultant is somebody who is old enough to have already made—or at least already seen—every imaginable mistake, but clever enough not to repeat one. The consultant comes to the client with forty years of scar tissue." In many respects, everyone in audio is in the same boat as the consultant, and the kind of frank and honest discussions of methods and procedures which appear in the Syn-Aud-Con newsletter can help to expedite the apprenticeship process. This is in marked contrast to the trade journals in which the system descriptions tend to be self-serving marketing efforts. Even the real turkeys are heralded as triumphs, and this is dangerous because it tempts others to step on the same land mines.

Early Patent for Sound Columns

The earliest patent, I know of, for sound columns is in the year August 1928 (assigned to RCA) and included stack and splay, as well. This is one year earlier than the previously quoted German patent issued in May 1929. 1989 is the 60th anniversary of the sound column.

1989-90 SYN-AUD-CON SEMINAR AND WORKSHOP

2-Day SEMINARS

New York Area
October 16-17

Washington, D.C.
October 26-27

Orlando, FL
November 15-16

Anaheim, CA
January 22-23, 1990

The on-the-road classes are a rapid fire review of audio and acoustic basics and their use and misuse in current system practices. Unlike our 3-day classes at our farm in Southern Indiana which are intended to develop specific audio skills in a limited number of basic audio tasks, our 2-day on-the-road classes provide a global approach to the areas of audio and acoustics that 40 years have shown us to be important to your success in this field.

Our 2-day classes are an excellent introduction to Syn-Aud-Con, the vocabulary of audio and acoustics, and an authoritative overview of what you may well be missing and need.

3-Day SEMINARS

The Farm in Indiana, 1990

April 26-28	August 23-25
May 17-19	September 20-22
June 21-23	October 11-13
July 26-28	

3-Day WORKSHOPS

Concert Sound Reinforcement
January 16-18, 1990
Chapman College
Orange, CA

Workshop Chairman: Will Parry
Maryland Sound

Workshop Staff:
Roy Clair, Clair Brothers
Albert Leccese, Audio Analysts
M. L. Procise, Showco
Mick Whelan, Electrotec

Facilities Coordinator: David Scheirman
Concert Sound Consultants

The workshop will be held "on location" with a live band so that the principles being discussed can be demonstrated. With the experience gained from last year's workshop, emphasis will be placed on those subjects that the exceptional group of attendees identified last year as vital to their interests, plus the insights the staff gained from having pioneered this type of workshop.

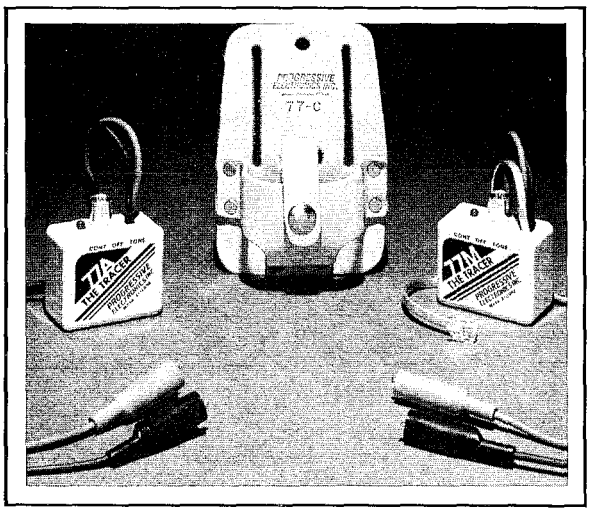
The workshop will be held at Chapman College, immediately before the NAMM show. We have a block of rooms in Anaheim for those who wish to combine the Workshop with NAMM.

Workshop Registration Fee
At the Farm - $500 for 3 days
On-the-road - $600 for 3 days

Seminar Registration Fee
At the Farm—$500 for 3 days
On-The-Road Classes—$450 for 2 days

Battery Operated Tone Generator

by
Bill Newton
Dallas, Tx



I recently tried a little battery operated tone generator. This is a good companion for the Music Supply TS-1 Test Set marketed by J. W. Davis Co. It uses a 9v battery, rugged, and costs about $25. It's really made for the telephone industry and includes continuity testing using an LED, or in the "off" mode can determine the tip and ring of a telephone circuit, if the line is powered. In the "Tone" mode, it sends out about a 3.7 vac open circuit, or about -1dBm (600 ohm load of course). It is equipped with good American made test leads and clips plus a choice of single or dual tones. The selector switch has a guard to prevent it from accidentally being turned on and letting the battery go dead. It is model 77A and manufactured by Progressive Electronics, Inc., 325 S El Dorado, Mesa, AZ 85202 (602) 966-2931. It's available from Specialized Products Co., Stock #367077, 3131 Premier Dr., Irving, TX 75063.

FSR RGB Sync Video Distribution Amplifier

The RGB-DA video distribution amplifier accepts an RGB sync signal from a video source and distributes it to three separate RGB sync devices.

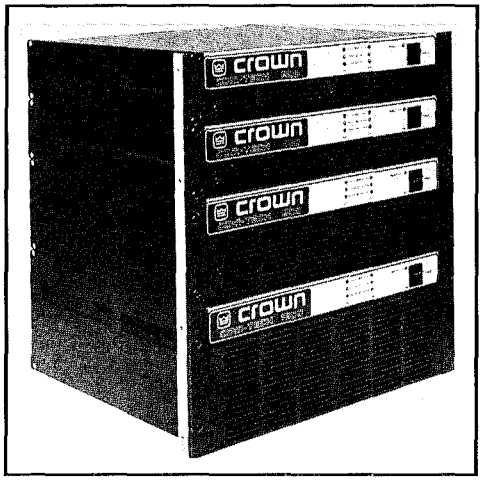
The unit has provisions for gain trimming + or - 2 dB from unity on the RGB signal, as well as a separate control for sync. All input and output connections are made with standard BNC connectors. A separate pluggable screw terminal connection is used to connect 24 volts to the unit.

Being part of the "EZ Box" line, the RGB-DA is packaged in a metal enclosure which includes a bracket for rear rack mounting.

Ever try to find a product like this? FSR is increasingly a highly useful resource to the sound and video system designer. Syn-Aud-Con's use of FSR products has shown us that they produce reliable accurate products that are trouble free and easy to use.

For additional information on the RGB-DA or the entire "EZ Box" line, contact FSR, Inc., 220 Little Falls Road, Cedar Grove, New Jersey, 07009 (1-800-332-FSR1).

The Crown IQ System 2000 Interface

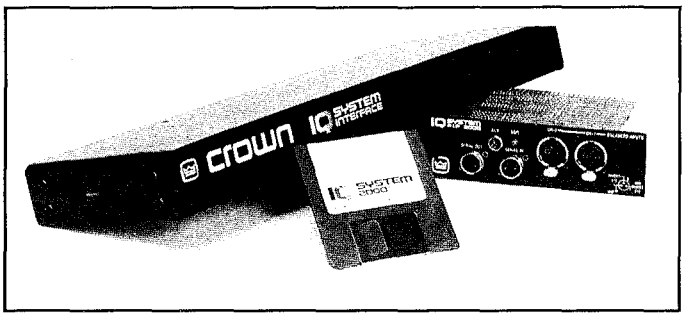


There are, at the present time, two major approaches to the use of digital technology in audio systems. One is to convert the signal into a digital format and only return to analog at the power amplifier. The second approach is to leave the signal in the analog domain but control its path, level, and supervision via a computer.

The Crown IQ 2000 system is an excellent example of the second approach. The TOA SAORI signal processor is an excellent example of the first approach.

The IQ System 2000

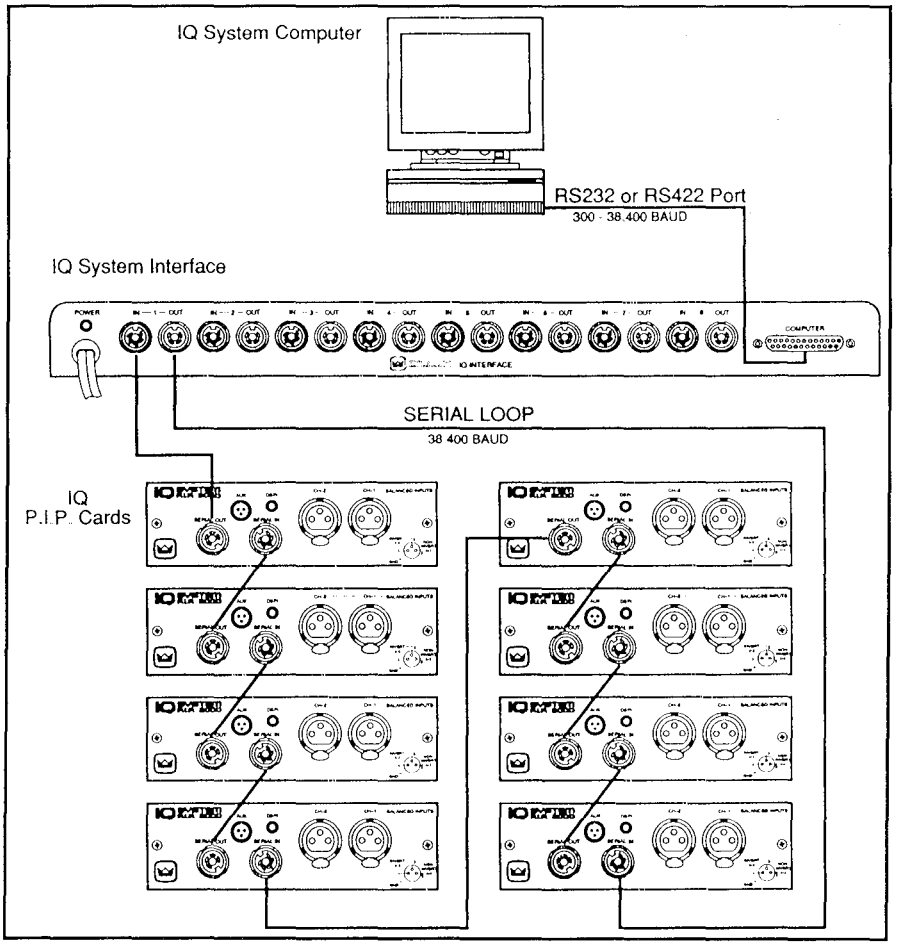
In our opinion, the 1990s will see the elimination of knobs on audio equipment meant to be seen by the customer. There will continue to be switches and knobs on the back of and inside components such as attenuators for initial set-up parameters but the operator-customer interface to the system will undoubtably become a computer.



In the IQ 2000 the computer is a MacIntosh. Using either an RS232 or RS422 port at 300-38,400 baud, the MacIntosh connects to the IQ System interface which in turn feeds a serial loop of up to 250 of the IQ-PIP (plug in panels) that plug into the rear of the Crown ComTech amplifiers. One IQ System interface can support up to eight serial loops, thus allowing 250 x 8 = 2000 power amplifiers to be under the direct supervision of one computer. It should be noted at this point that the level of supervision the computer has now is but the tip of the iceberg compared to what it can have with these same components in the future.

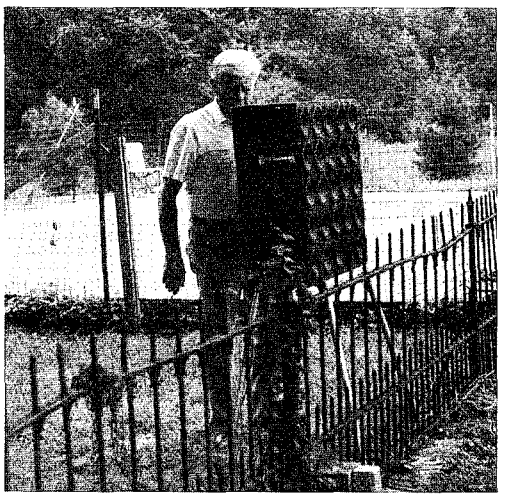
Currently the computer can control input level, polarity, on and off (complete with sequencing) as well as monitor output level, the input-output comparitor (IOC) condition, and indicate where system faults are located.

As was noted by an observer seeing 100 lawyers at the bottom of a lake with their feet in cement, "Its a good start". Software and the inclusion of additional sensors for the three computers to watch and manipulate will eventually complete the system.



IQ System 2000

July Class at the Farm



Special instructors this summer were Farrel Becker, Helmuth Kolbe, Gene Patronis, Mary Gruszka and Kurt Graffy.

Special features have been the 25 speaker Bessel array with each of the 25 speakers in its own separate enclosure within the main enclosure and the "In-The-Ear" ITE™ recordings played back over the "Pinna Acoustic Response" PAR™ playback system. One of the more interesting experiences is to sit outdoors and hear a reverberant concert hall surrounding you.

Our current use of the Sony TCD-D-10 Pro DAT recorder has allowed remarkable advances to be made in our ITE recordings. Prior to the acquisition of the DAT quality duplicates were not possible.

The 25 speaker Bessel allows rapid ear calibration relative to the sounds produced by both interfering arrays and non-interfering arrays.

Ernie and Vivian Pence have now brought more of the house into useability including a workbench in the north room for staff use before classes. (For those of you who attended the farm classes this summer, we have gone so far as to get a quote on re-siding the farm house.) During hikes this summer we have seen the usual deer, coyotes, racoon, hawks, wild turkey, ruffled grouse, and even a mink. Our horses continue to prosper and are occasionally visited by the neighbor's horses after they stroll through their electric fence. Princess and Patch have become like mother and daughter and we've rarely seen two happier animals. The four cats have had their best summer though Pete recently complained to me that Tillie had unexpectedly "mugged" him out in the garden. There are now two goats at the farm. TuTu has been joined by Bo and after two weeks of butting heads and other social amenities, now eat out of the same dish.

Don turned 61 in September, and and Don and Carolyn observed their fortieth wedding anniversary in the same month.

As a result of the reception given the farm classes, we are scheduling a full set of them for 1990 beginning the end of April (mushroom season). We will limit attendance so get your reservations in early as there will only be one class per month. We hope all of you will get a chance to attend one and experience the Syn-Aud-Con farm.

Random Thoughts on PAR Recordings

We now have overwhelming evidence both from our own recordings and from the current literature in the *Journal of the ASA* that there are indeed superior listeners in terms of the signal that arrives at the eardrum. By superior, we mean both for speech intelligibility and for music quality.

We now are making PAR recordings with the new super quiet Etymotic ITE Mark I microphones recorded into a new Sony portable TCD-D10 PRO DAT recorder. The PAR playback system consists of four UREI 809 Time Align™ monitors powered by IRPI and Techron amplifiers. The front end is a pair of Allen Burdick's Quad amps modified for PAR use. The result is absolutely spectacular recordings. All of this experience spurs the following comments.

1. In our opinion, a two-channel, four-loudspeaker playback system allows the most realistic cancellation of front speaker crosstalk interference when "in-the-ear" ITE recordings are properly made. At the suggestion of Ted Jones of Indiana University, we placed the loudspeakers on the floor and angled them upward. We found that keeping all devices (loudspeakers, electronics etc.) on the floor and keeping the ear and eye level area unobstructed still further improved imaging especially in the front of ITE recordings.

2. We now have overwhelming evidence both from our own recordings and from the current literature in the *Journal of the ASA* that there are indeed superior listeners in terms of the signal that arrives at the eardrum. By superior, we mean both for speech intelligibility and for music quality. Some eardrums receive higher level signals due to the size of the pinnae, configuration of the ear canal, shape and size of the concha, and size and shape of the head.

3. It would appear that having marked dissimilarity between left and right ear structures to the eardrum provides some advantage to the dual hemisphere processing in the brain. This seems particularly true in terms of directional perception skills.

4. In attempting to make dubs from our older Sony analog cassette unit we found that it was impossible to preserve the high frequency information (i.e., the pinnae comb filtering) due to the difficulty of achieving phase alignment. The DAT solves this problem with ease. As a consequence, we believe that this problem has, in the past, more than likely, blocked researchers from making progress because their recording instruments lacked accuracy in a critical frequency-phase area. We remember that Doug Jones had the same difficulty when he tried to duplicate his LEDR tapes.

5. We are currently researching the idea of using two PZMs 15 to 20 feet apart in a more or less conventional spaced manner (the Häas distance apart) mixed in with the ITE channels for the elimination of front loudspeaker crosstalk interference.

At the present time (August 10, 1989) we feel that we are matching the quality of the very best commercial CDs and surpassing them remarkably in terms of spatial imaging.

The Next Step

We believe that the next step is to set up a screening system to locate superior listening heads, have them measured, and their ear canal molds made. We will then use them as the model for a prototype artificial recording head with the same microphones we are now using permanently mounted in the pressure zone of the eardrum. One way to accomplish this would be to record, with the heads to be screened, speech intelligibility tapes in a difficult location and then see which "head" allowed a separate set of listeners (on headphones) to make the best score on the recorded material. Live listener scores and their standard deviation would, we believe, match the spread we observe in using different listening heads. That hypothesis alone makes such an endeavor worth while.

As a result of all of the above, we sincerely believe that we are on the threshold of a new recording technology that will allow remarkably improved archival records of great performances.

Using Constant Voltage Distribution

The term constant voltage is applied to systems that do not vary in output as additional devices are paralleled on its output. See Fig. 1. This does not mean that the voltage doesn't vary—it does with the source signal—but only that adding or subtracting devices does not effect the source signals' voltage variations.

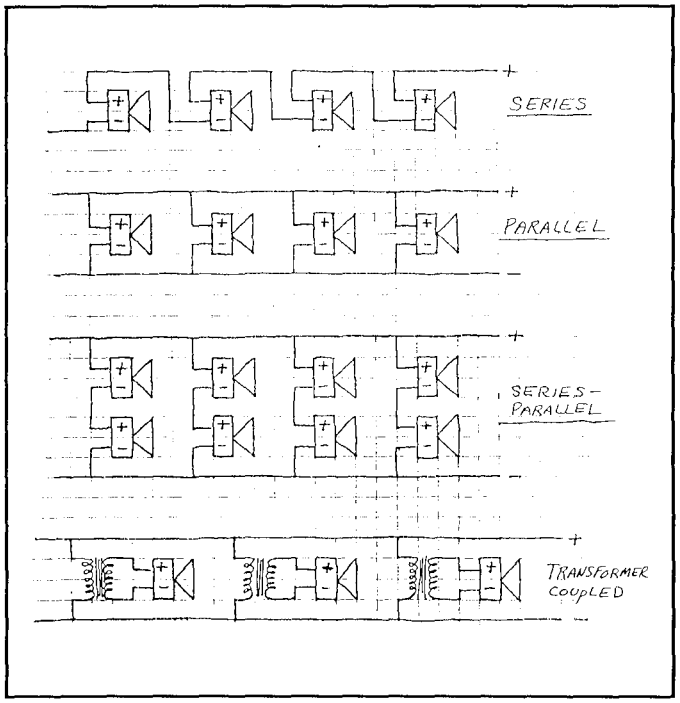


Fig. 1-Loudspeaker Wiring

Common versions of this technique are 70.7 volts, 25 volts, 100 volts, and 200 volts. This technique normally utilizes transformers (Fig. 2 A & B) and then quality often is the determining factor in the systems quality (Fig. 2). In wiring home entertainment distributions systems the 70.7 volt system is usually the best choice (25 volt is primarily used in wiring school intercoms, 100 volts is used in Europe, and 200 volt systems are used for very large systems with very long wiring runs). In choosing 70.7 volt transformers we compare the specifications against the ideal transformer.

The ideal transformer neither stores nor dissipates energy, but in order to be ideal, a transformer would have to meet the following three requirements.

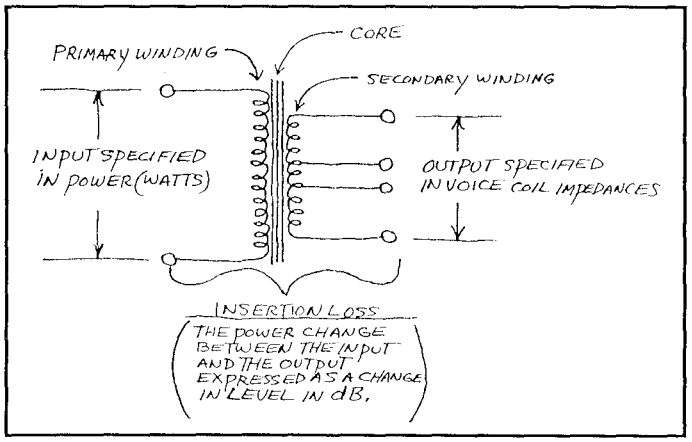


Fig. 2A-Constant Voltage System Transformers

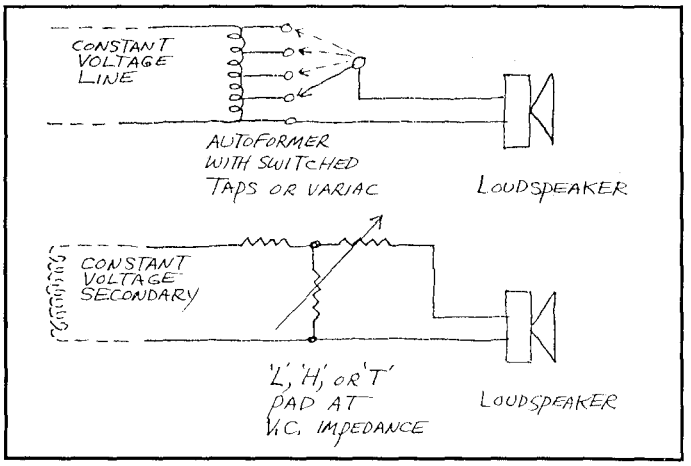


Fig. 2B-Controlling Levels in Constant Voltage Circuits

(1) Have infinite primary and secondary inductance, but a finite ratio of primary to secondary inductance.
(2) Have perfect coupling between primary and secondary.
(3) Have no resistance in primary and secondary windings.

The first condition is satisfied by using cores of extremely high permeability and a great number of turns on the coil. The second is approximated by arranging the coils mechanically so that the leakage Flux between turns is very small.

The third condition is satisfied by the fact that for audio frequencies the resistances of the windings are small compared to their impedance in the frequency ranges under consideration. See Fig. 3.

For all practical purposes with good iron core transformers the ratio of

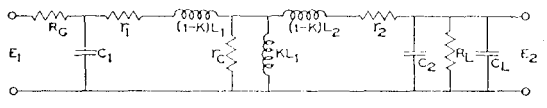
$$\frac{L_1}{L_2} = \left[\frac{N_1}{N_2}\right]^2$$

also

$$\frac{E_1}{E_2} = \frac{N_1}{N_2} \text{ and } \frac{I_1}{I_2} = \frac{N_2}{N_1}$$

It is now possible to set up the formula for an equivalent circuit for an audio transformer, since in the practical transformer we have resistance in the coils which is the equivalent of a series loss, distributed capacities which represent shunt reactances on both primary and secondary, and losses in the magnetic circuit which look like resistance shunting the winding.

The circuit is as follows:



C_1 = distributed capacities of primary
C_2 = distributed capacities of secondary
C_L = distributed capacity of the load
K = coefficient of coupling
L_1 = inductance of primary
L_2 = inductance of secondary
r_1 = resistance of primary
r_2 = resistance of secondary
r_C = exciting current loss (core loss)
R_G = generator resistance
R_L = load resistance

Equivalent circuit of audio transformer

Figure 3

The condition most likely to be encountered is excessive, frequency dependent, insertion loss, usually low frequency. The lowest impedance the line should be allowed to drop to at any frequency is:

$$Z_{MIN} = \frac{5000^*}{W_T}$$

*E^2

That is the voltage squared divided by the total power W_T.

Some transformer manufacturers compensate internally in the transformer windings for the insertion loss so that the specified power at the input is delivered to the output (i.e., they draw more from the line by the amount of the insertion loss, than specified). Others merely deliver less power, as indicated by the insertion loss, to their secondary.

Some Useful Equations

To find the primary impedance Zp of a transformer

$$Z_P = \frac{(E_{OUT})^2}{W_S}$$

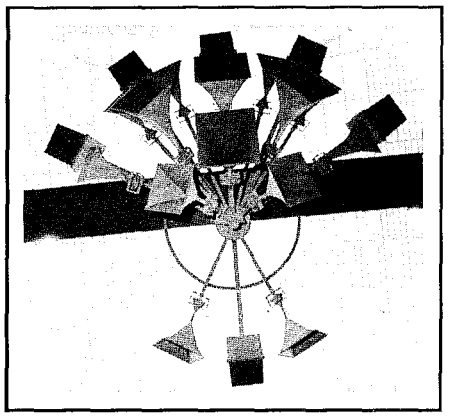
Where: W_S is the loudspeaker power in watts
$(E_{OUT})^2 = 5000$ in the case of the 70.7 volt system.

The total impedance of the line is

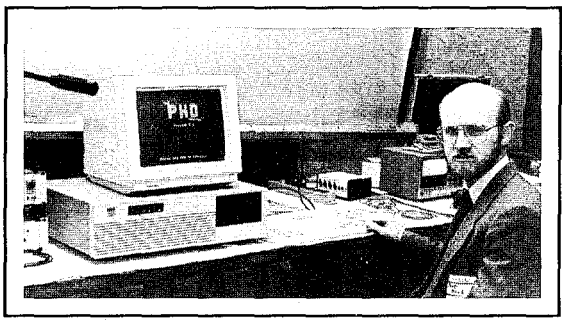
$$Z_T = \frac{1}{1/Z_1 + 1/Z_2 + \ldots 1/Z_N}$$

and the output voltage is found by:

$$E_{OUT} = \sqrt{Z_P W_S}$$



Uniquely Beautiful Cluster

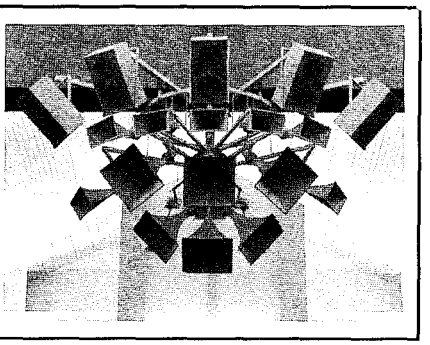


Art Hiebert demonstrating the PHD Program to the Vancouver Class

David Ruff and Art Hiebert of Unique Communications in Calgary attended our Vancouver seminar last February. Art has had a lot of experience with the PHD program so we asked him to demonstrate it to the class. He did a beautiful job.

Soon after the class in Vancouver we read an article written by Art in Sound & Video Contractor (March 20, 1989), *Church Sound System Doubles as Modern Art.* The sound system is truly a work of art. It is rare to judge a cluster on esthetics, but this one certainly enhances the visual aspect of the building, which is modern high-tech sculpture.

It is one thing to make the work-of-art cluster look good, but making it sound good required genuine engineering: very careful attention to signal alignment and structural engineering.



Briercrest Bible College, Caronport, Saskatchewan, Canada. The sound system was installed in 3,800 seat Hildebrand Chapel complex. The main cluster consists of six EV TL606DX bass speakers, 15EV HP horns and 15EV DHIA drivers. This is a view of the cluster as seen from the back of Hildebrand Chapel.

The Null Frequency Interval (NFI)

If, in examining a 1/3-octave real-time analyzer plot, you find that you have a deep notch at 1500 Hz, most likely it's the first notch in a comb filter series of notches. What distance mis-synchronization would you look for?

Since the null frequency interval NFI is directly proportional to velocity c and inversely proportional to distance d then:

$$NFI = \frac{c}{d}$$

$$\text{therefore, } d = \frac{c}{NFI}$$

If we make c in inches per second (i.e., 1130'/sec x 12" = 13560"/sec) we would find for our question above that:

$$d = \frac{13560}{1500} = 9"$$

It is a truism that the most dangerous time for authortarian regimes comes not when oppression is worst but when expectations are rising.

Mentors

What does a "mentor" look like? What do they do? Why aren't there more of them?

A "mentor" looks like the man on the left in the photograph. What they do is supply a support that is unshakable to pilgrims trying to know the truth. There are, unfortunately, only a limited number of men and women capable of being mentors because it takes true intelligence coupled to academic training (two quite different properties) in an individual who loves his fellow man as God intended we all should.

We salute Gene Patronis a model mentor and a wonderful friend to both Carolyn and myself, and to the Syn-Aud-Con concept.

Ratios: Grass-to-Weeds Ratio & Others

As I walked through one of the pastures on the way to the woods with the dogs, I suddenly realized that I was as likely to be occupied with the grass-to-weeds ratio, G/W, as I was the signal-to-noise ratio. Then I asked myself, "why grass-to-weeds, why not weeds-to-grass?" I answered myself that S/N really is useful/unwanted; therefore grass-to-weeds is the correct expression. We have a cat-to-people ratio of 2/1; a dog- and horses-to-people ratio of 1/1, and a goat-to-people ratio of 1/2. Carolyn and I have decided that a summer at the farm is a happiness ratio of ∞/1.

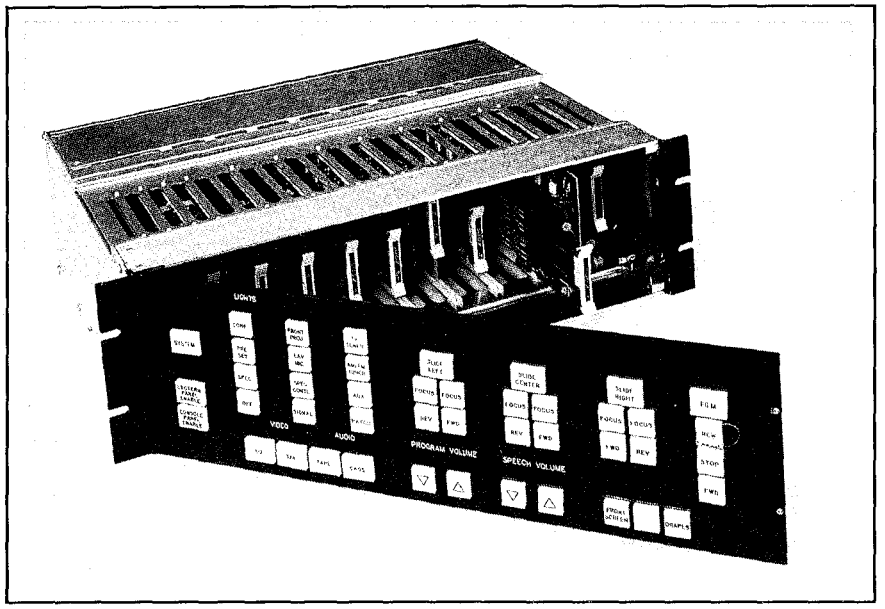
Gerry & Clay Barclay

Syn-Aud-Con is always pleased to hear competent people are appointed to jobs that will exercise their skills.

Both Gerry, and her husband Clay, are extraordinary people that we are delighted to number among our friends. We wish them both the best of success in their new endeavors.

TSI Special AV Products Division

AV can intimidate many people, especially the small sound contractor. He or she may fear the complications that might arise if they purchased AV, and have stayed away from it. Mario Maltese, CEO at TSI, is a very special person. We have written a lot about him in Syn-Aud-Con Newsletters and Tech Topics, and we will be writing more. Mario was a member of the staff for the Computers In Audio workshop. He shared all of his programs so freely. It was like he was saying, "why should you have to spend all your time working out these programs when I have already done that work." That is the philosophy behind the TSI Special Products Division, 393 Jericho Turnpike, Mineola, NY, 11501, Ph. 516-294-5390.



Syscon 200 audio—visual control system

TSI started a Special Products Division because they once were very small and know how difficult it is to take the first baby steps to enter a new field. That is where they were ten years ago. Now they are 60 employees and 10 full CAD stations.

TSI, over the years, developed many special products that helped them meet their clients' needs.

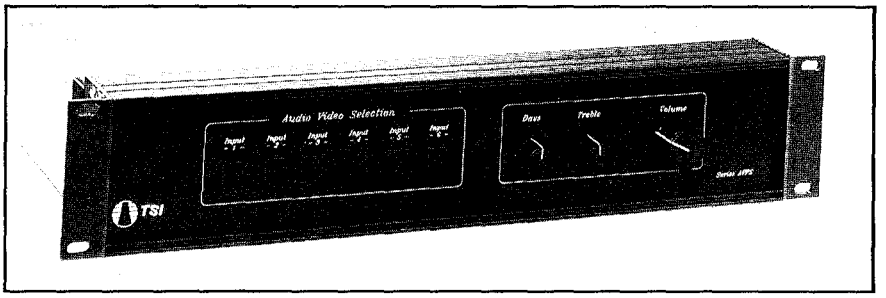
The Special Products Division was formed, not just to share those products with other sound contractors but to make them available as a system. TSI products are for the sound contractor that has a very special need for a one-of-a-kind system, with the stress on **system** rather than components that could be assembled by a manufacturer. TSI is filling a special slot, not taking anything away from current manufacturers but adding a service to the industry.

There are a lot of good, professional contractors out there, contractors who have proven themselves to their customers. It makes good business sense to diversify into other markets, especially growing ones. It makes poor sense to go it alone.

TSI is quick to align itself with the mission of the small contractor to get the job *done*. TSI doesn't just want to sell parts, they want to be part of the team that installs them. They want to help the contractor to succeed in AV.

TSI Special Products Division provides complete systems, fully documented, with all the assistance the contractor needs to let them do AV systems. The sound contractor can give their requirements to TSI and they will configure the system for them. A **complete** AV control system is delivered rather than individual products.

They provide the hand-holding that gives the small contractor the courage to enter this exciting field.



AVPS-6 audio—video switcher

The Arrogance of Man

In the evening just before retiring for the night, I walk with the two dogs up the lane to the laboratory in the old house. I can still see well in the pitch dark, and I re-test myself every night by trying to spot Patch, our German shepherd who is black, as a bulk in the night as she comes and goes chasing whatever is at hand, or should I say tooth and paw. Sometimes it is so dark you can't see even your own feet, and other times, such as full moon, one can read a newspaper.

The stars often form a deep vault over our heads, and one has only to raise one's vision to feel how insignificant man is. It's harder to see infinity in the city; not impossible, just harder. The countryside is full of security lights and there's one in front of the old house, put there before we had a say in the matter. Our new house and the office have no such pollution, so I'm able to walk to the old house and out to the barns in a wonderful mantle of darkness. I carry a flashlight, but rarely use it. If the dogs pursue something directly toward me, I take a quick look with the light as I'm not interested in contesting the right of way with a skunk.

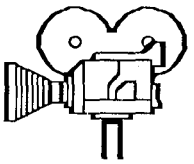
Snakes are rare and at night sluggish, and it's amazing how their length pops out to your vision even on a pitch black night. Near water and back in the deep woods, I would not be so casual about snakes at night, but along the lanes there is little to fear.

When one walks where wild things live, it should be without fear or aggression. Where you are is their home. They'll be here in perpetuity long after my footprints are blown away by the winds. One properly walks in alertness and readiness: alert to the wonders and ready to counter the aberrant (wild dogs, rabid skunks etc.)

There are no colors at night, just forms, shapes, and sounds. Night sounds are quite revealing as to who and what owns the majority of mother earth. Man is largely an intruder and those of us who just want to be witnesses must, of necessity, stay out of the way of the native—animal or human.

It does, however, always strike me as absurd to think that anyone ever really possesses the land. One can work the land or ruin it, but it's going to outlast us and many like us. We are the temporary custodian at best. I honestly believe that no man can be fully free while in the sight of another, but in the final analysis, freedom is mental and space merely the catalyst to consider it.

I Wish I'd Had a Camera



In years gone by, really bad papers were detected by watching Paul Klipsch roll up the whites of his eyes at the meetings where they were given. Those papers weren't printed. Harry Olson, in his years as the editor of the *AES Journal*, required that the writers knew what they were talking about and he personally, in one case (Heyser's first paper), saved great papers from being rejected. Often papers are subjected not to peer review but to poor review.

I Wish I'd Had a Camera

I used to tell Syn-Aud-Con classes about Paul Klipsch rolling back his eyeballs. I am sure that everyone, who didn't know Paul Klipsch, thought I was taking poetic license. I wish I had had a camera at AES about 12 years ago when we were talking with Paul while a group stood about, as they always did when Paul Klispch was present. Ron Wickersham, of Alembic and Pressure Recording Process fame, was standing at Paul's elbow when suddenly you were looking at the whites of his eyes. We all laughed out loud, not at Paul, but at the look on Ron's face.

An Intelligibility Experience

After we had received our newer and quieter probe microphones for ITE recordings, we took advantage of a rehearsal of the Bloomington Symphony Orchestra to try them out in Ted Jones' ears. The soloist for the concert was I. U. Professor Emeritus Fritz Magg, a world renownd cellist. In the process of recording the rehearsal, we made a startling discovery; namely that intelligibility dropped noticeably when the ITE microphones were taken from Ted's ears and placed in Carolyn's. With the ITEs in Ted's ears you can understand what Fritz Magg is saying—German accent in a highly reverberant space. With them in Carolyn's ears you cannot. (We are not talking about hearing acuity as far as frequency range is concerned.)

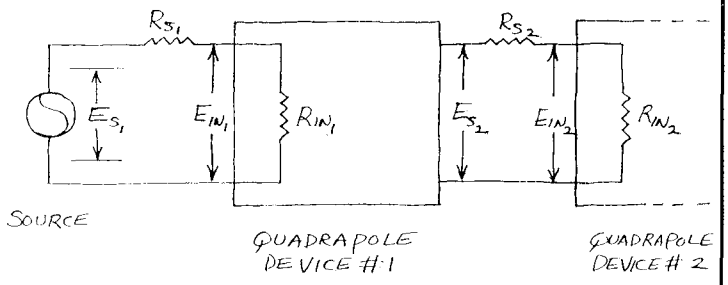
As a result of this and a few other experiences of this sort, we believe that there may well be correlation between the standard deviations in live listener tests for intelligibility and the differences in head size, shape, pinnae configuration and ear canal differences from ear-to-ear.

Once Again—Levels

The dBm and its use has nothing whatsoever to do with 600 ohms, balanced circuits

We continue to see published calculations of power in a load at the input of a device when what is called for is the available input power from the preceding device. Lest we be misunderstood, we did not invent the concept of available input power AIP. Generations ago really sharp electrical engineers did, and it has served well ever since.

The Gain or Loss of a Link Circuit Device
Circuit may be balanced or unbalanced (as shown)
R_S may be any value equal to or less than R_{IN}



The Gain of the Quadrapole Device #1

$$\text{GAIN IN dB} = \left[10\text{LOG}\left[\frac{(E_{S_2})^2}{0.001 R_{S_2}}\right] - 6\text{dB}\right] - \left[10\text{LOG}\left[\frac{(E_{S_1})^2}{0.001 R_{S_1}}\right] - 6\text{dB}\right]$$

I read with astonishment the totally ignorant statement, "In fact, about the only time that "dBm" is appropriate these days is when the equipment is driving very long cables, like those used in large sound reinforcement systems or multi-studio broadcast production complexes, where true 600 ohm balanced circuits are essential to reduce susceptibility to induced noise and high frequency losses."

To paraphrase *My Fair Lady*, "Damn, damn, damn, damn—I've not grown accustomed to their voice."

The dBm and its use has nothing whatsoever to do with 600 ohms, balanced circuits or anything else they quote. The dBm is appropriate when you want to state the power level, nothing more—nothing less. Writers who don't know A of ABC should turn in their pen and stick to press releases, advertising, and other forms of misinformation.

Available input power is:

$$\text{AIP} = \frac{(E_S)^2}{4 R_S}$$

and the level of AIP in dBm is:

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

Only at the final load is the level of output power calculated:

$$L_{\text{OUT dBm}} = 10 \text{Log} \left(\frac{(E_L)^2}{0.001 (R_L)} \right)$$

The figure illustrates the definitions of AIP and how to compute all of the parameters necessary.

Acoustic, Audio, and Velocity

Acoustic frequencies travel at the velocity of sound. This velocity varies from in air at 344.42 M/s to water, salt, 21°C at 1520 M/s. Sound travels through glass at 5200 M/s, steel at 5050 M/s, and concrete at 3400 M/s. Sound travels through the human body at 1558 M/s.

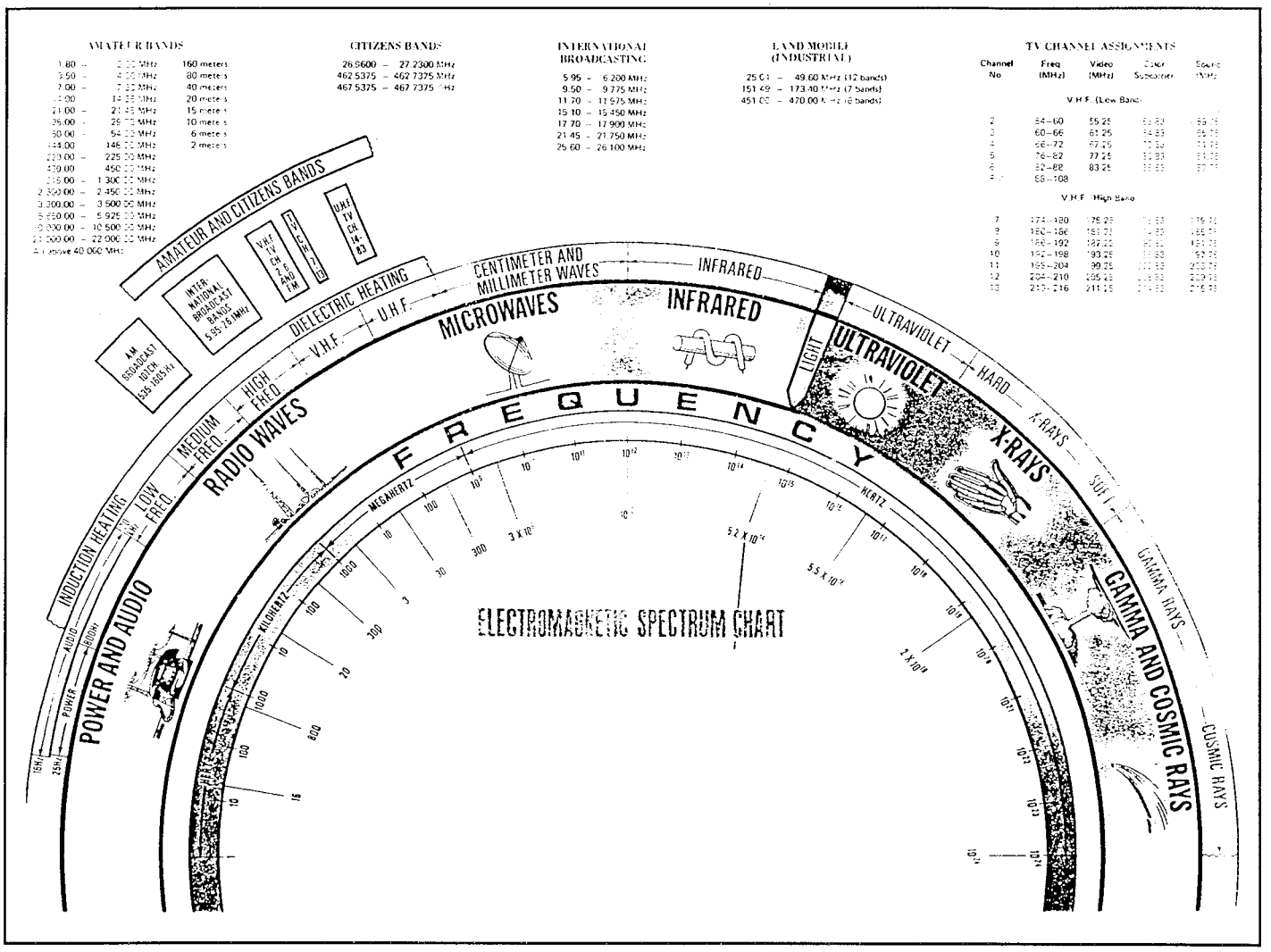
Audio frequencies theoretically travel at the speed of light, 186,000 mi/sec. In actual lengthy circuits such as voice frequency loaded cable (14,000 miles/sec) on 19 gauge cable with H88-50 loading compared to about 150,000 mi/sec on high quality carrier circuits), we find other velocities and as in acoustic propagation frequency dependent as well.

When the word "audio" is used at Syn-Aud-Con, it is assumed that some form of the electromagnetic spectrum is under discussion with its attendant velocities and that when the word "acoustic" is used that acoustic velocities are assumed. Because the wavelength λ is found by:

$$\lambda = \frac{c}{f}$$

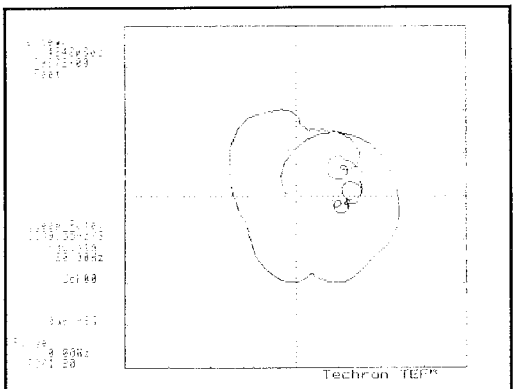
where c is the propagation velocity and f is the frequency, some quite strange λ answers can occur if the wrong velocities are chosen.

If we use the language of science with care, examining in our thoughts what we mean when we speak, then other properly trained persons will be able to understand us.



The electromagnetic spectrum

DeltaMax™ (A Need Met)



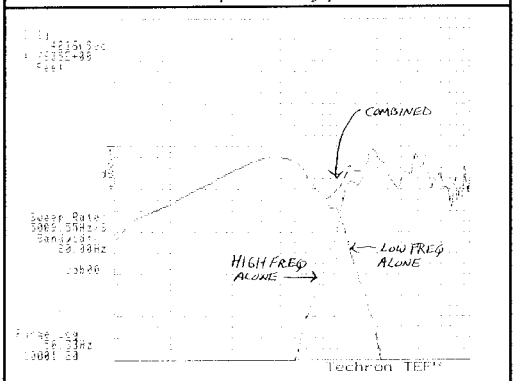
An exceptional Nyquist

Hellmuth Kolbe of Zurich, Switzerland helped teach the July class at the farm. Todd Rockwell and Tom Gallagher from EV were in attendance and demonstrated the EV DeltaMax DML 1152 units. to the class; and, we had an opportunity to make TEF measurements.

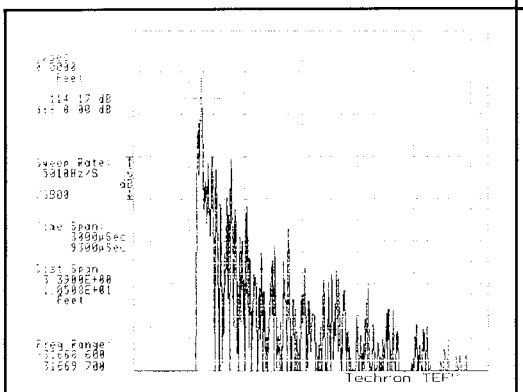
We were impressed. For their size and intended use their polar characteristics are excellent. Their front-to-back ratio is never worse than 12 dB even at the lowest frequencies. This is very important as so many microphones are worked directly under the loudspeaker. If you have a sound system that has feedback problems in the 125-500 region, check the level of the signal from the back of the loudspeaker.

The DeltaMax system includes an electronic controller which includes a pre-set crossover frequency (1,100 Hz.), high-frequency CD horn EQ, analog low-frequency delay and low-frequency step-down EQ in addition to protection circuitry. The alignment between the woofer and high frequency unit was out by 332 usecs. We were told by Electro-Voice that this was the result of engineering the system to have the vertical lobe at crossover centered at 0 (perpendicular to the cabinet). The woofer doesn't quite reach the high-frequency unit in the crossover region.

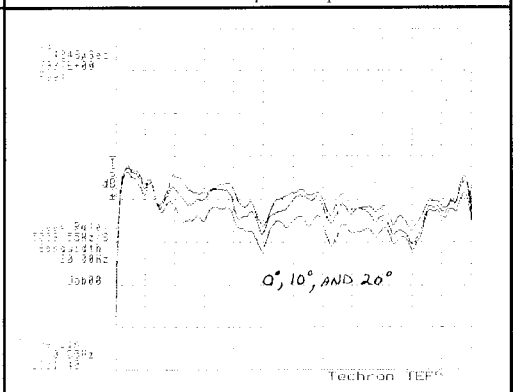
The DeltaMax includes protection circuitry, which monitors the excursion and temperature of both the bass and high frequency drivers as well as amplifier clipping. If the electronics sense an overload situation at the amplifier terminals, the input signal is lowered thus leaving the spectral balance intact.I would make the following comments. This is the best unit for its size that we have tested to date that can handle the kind of power required, control its pattern and not have distracting distortion. It is well packaged and would be our first choice in its size category.



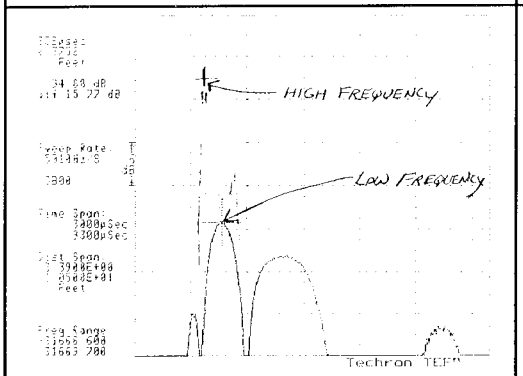
EFC of low and high frequency units alone and combined. Often we are advised in the literature to flop polarity at the crossover region to smooth the frequency response. Indeed it does, but it completely fouls the phase response, which sounds worse than the dip in response.



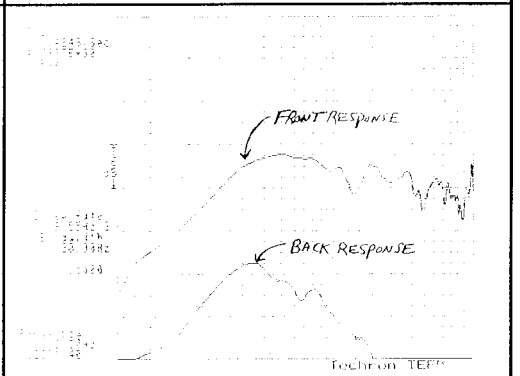
ETC of the DeltaMax



EFC 0, 10 and 20 degrees off axis.



ETC of woofer with the midrange and tweeter superimposed.



Front and back response showing never less than 12 dB separation.

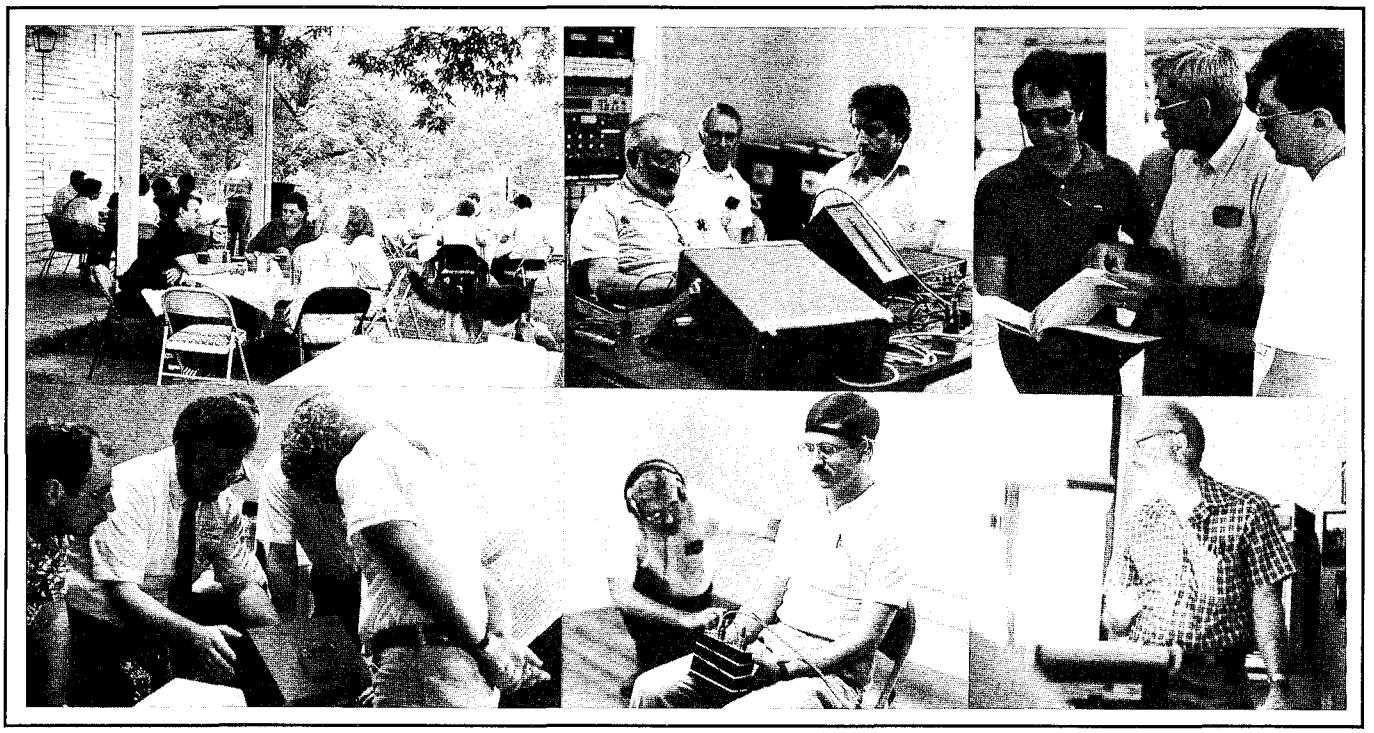
Synergy— the August Class

When people ask what the "Synergetic" in our name means, we can't help but think about events like the August class at the farm.

Synergy in chemistry is when an alloy's strength exceeds that of the sum of the strengths of the ingredients (i.e., 2 + 2 = 5)

As you can see from these pictures, the participants came together to both study and have fun. It was a full class and a class full of persons prepared to receive what Gene Patronis and I had to share.

These classes tell us that the one instructor for every six people and the extensive hands on is what they have been looking for. Most are surprised at how scenic our part of Indiana is. An increasing number are taking advantage of their visit to the farm to see Lake Monroe, the largest man-made body of water in Indiana located in the middle of the 40,000 acre Hoosier National Forest; Nashville, Indiana with its folk art, theaters and restaurants; and Spring Mill State Park with its grove of virgin timber and the remarkable Pioneer Village. Bedford's limestone quarries are world famous and a good deal of Washington, D.C. came out of the ground in Lawrence County, Indiana.



SwifNet - the Instant Network

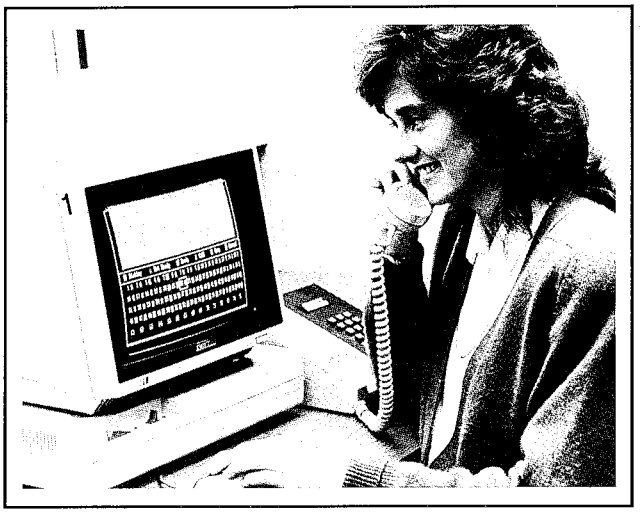
If you know anyone with a need to handle thousands of incoming calls in the space of a few minutes without losing hundreds of them or costing a fortune, you should tell them about SwifNet

The picture reproduced here of John Prohs, Richard Heyser and V. M. A. Peutz puts John Prohs in very august company, but that is where John Prohs belongs. He is a very creative man and very much at home in the company of talent.

Most of us will recognize John Prohs' name as the "P" in the PHD computer program (Ambassador College, which has donated the proceeds from the distribution of PHD, which raised over $35,000 for the Heyser Scholarship fund).

John also heads an engineering team at Ambassador College (Worldwide Church of God) who developed SwifNet.

SwifNet is an intelligent call-extending system that instantly establishes and manages a network of remote agents. The agents can be in the same room as SwifNet or thousands of miles away. SwifNet has routed more than 2 million calls to workers at home throughout the nation.When John told AT&T what he wanted built, they told him that it couldn't be done. So John did it!

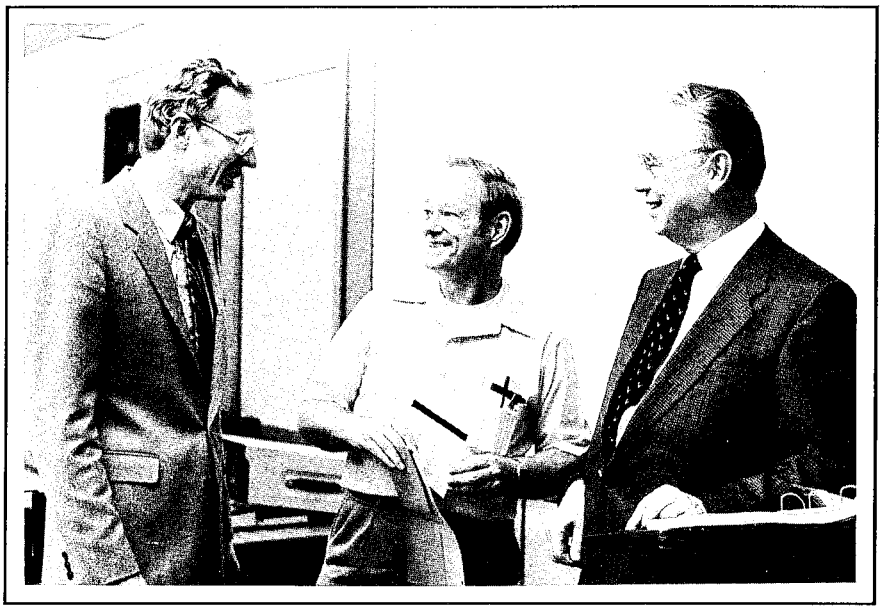


A Supervisor at a control center in Pasadena at Worldwide Church of God

SwifNet was developed to handle the increasing telephone response to the World Tomorrow telecast produced by the Worldwide Church of God. Expanding the call-processing capability of the Church's call centers would have required establishing additional call centers. Instead of incurring this large expense, the Church sought help from Ambassador College, its educational affiliate. That is when John and his engineering staff entered the picture.

We personally watched SwifNet in operation in Pasadena, the home of Worldwide Church of God. Over 1,400 calls were handled in less than 10 minutes from the control point in Pasadena but the phones were actually being answered in volunteer homes all over the United States. A supervisor at the control center in Pasadena can monitor any call, can assist on any call at the push of a button, has a record of the number of calls answered and drop-offs. The system is completely secured from outside intrusion.

When one sees the sophisticated nature of SwifNet, it is easy to understand how PHD has developed under John's care.



L. to R.—John Prohs, Richard Heyser and V.M.A. Peutz

Speech Intelligibility Workshop II May 1990

We will convene a second speech intelligibility workshop in the spring of 1990 under the auspices of Indiana University and Synergetic Audio Concepts because:

1. In-the-Ear, ITE, technology has advanced to the point where reproduction of intelligibility, or lack of it in difficult environments, can now be recorded with sufficient accuracy via DAT to allow meaningful playback at a later time.

2. The TEF analyzer now measures %ALcons, RASTI and full STI .

3. Newcomers to audio need a reference recording that lets them hear the difference between 5%,10%, 15% and 25%ALcons under conditions where the cause is (A) noise, (B) early reflections, or (C) reverberation.

4. Since the first intelligibility workshop we have found, through ITE recordings, that there are differences in the listeners' outer ear configurations that may be sufficient to explain the major portion of the standard deviations seen in the statistical analysis of live listener groups.

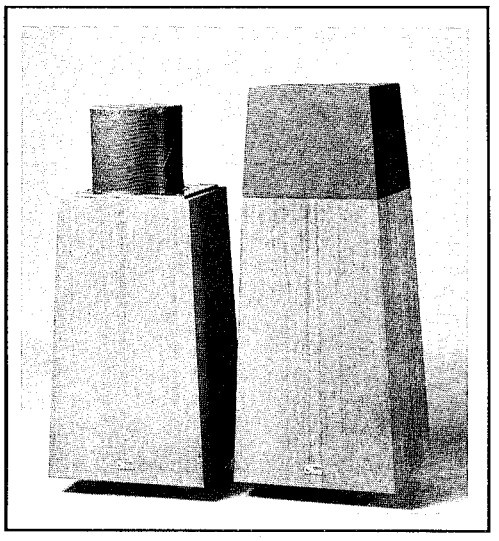
Workshop Plan

We intend to use at least five different listeners, selected for major variations in head and outer ear configurations, to make ITE recordings in at least three different acoustic environments. At the same time we will make normal live listener group tests with word lists plus full TEF intelligibility measurements using the measuring microphone and In-the-Ear, ITE, measurements. Each environment will be both recorded and measured at a series of locations within each of the three environments.

The same source, a 25 speaker Bessel array, will be used in each recording and measurement to ensure that the differences heard are either listener or location dependent. The Bessel can be varied from the Q of a single cone to a highly directional array.

One of the true benefits of the first intelligibility workshop was the chance the participants had to calibrate their own hearing. This workshop once again offers that unique experience under the guidance of real authorities in the field.

Ohm Loudspeakers



Choosing personal loudspeakers is a difficult task, at best. What you like on short exposure may grate on you with extended listening. Few wide-range loudspeakers sound good while listening to the news on FM radio.

We have been fortunate to spend the larger part of the summer listening to Ohm-Walsh 5s. Since having them on loan during our 3L Workshops last summer we have yearned to own a pair.

The Ohm Walsh-5s may have shortcomings, but so far, we are unable to detect them. They simply do wonders with Dorian recordings and make whatever source we choose sound smoother, cleaner, and less like loudspeakers than any other system we have had the pleasure of hearing.

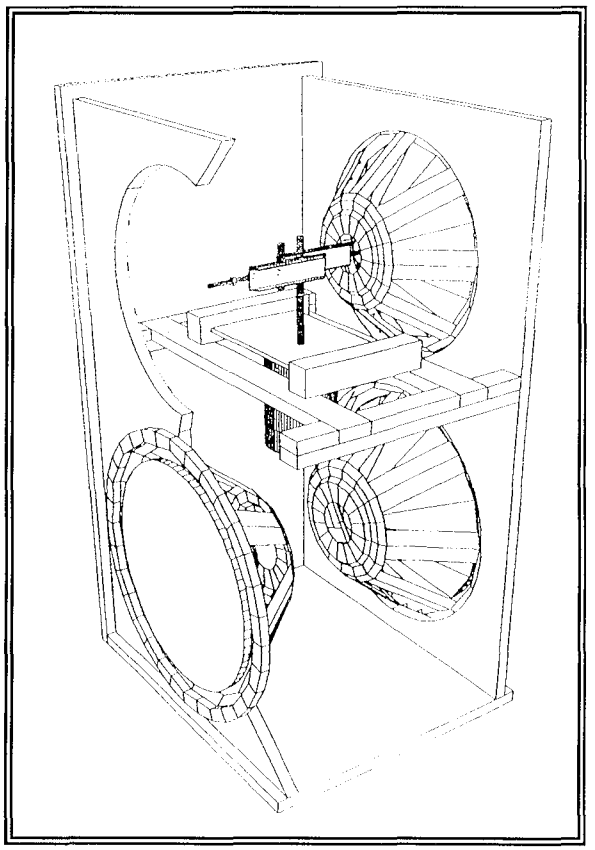
Now, I can startle you with the complete illusion of a live event with our four Time Align UREIs with our new DAT recorder and ITE recordings—they're truly mind boggling.

But, to go from the lab to the house and play normal recordings and not suffer catastrophe effect makes the Ohm-Walsh 5s absolutely fantastic. Listening fatigue simply fails to materialize when they're used. If you love classical music, as we do, seek out an opportunity to listen to Ohm-Walsh 5s on an extended basis. The remainder of your money can then go to Peter D'Antonio to make the room worthy of the sources.

Intersonics Contra Bass Servodrive Subwoofer

The Intersonics Contra Bass Servodrive subwoofer is a clever and effective audio component. We are currently feeding both channels from our PAR playback system into one of them. We have made a DAT ITE recording of an exceptionally talented percussionist at Indiana University which provides a full measure test signal.

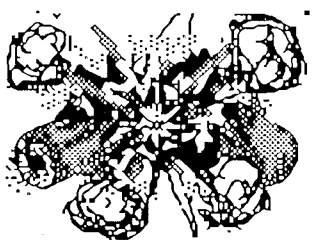
I had tended to think of subwoofers as a "Johnny-one-note" used in discos. I simply hadn't realized how vital it is to cover the bottom two octaves smoothly (16-32 and 32-64Hz). This unit has accomplished just that. The Intersonics subwoofer is being used very subtly to provide the identical levels encountered by the live listener at the live performance. Interestingly, as little as 35 watts to the unit does the job. Its 2-3 ohms nominal impedance requires a good stable amplifier.



The servo mechanism is based on a special high-tech, low inertia motor. Unlike a voice coil when conventionally suspended that can become non linear with large motion, a motor can provide unlimited motion or rotation. The rotation in either direction is exactly proportional to the input signal voltage and current. As a result of this tight electrical coupling, it is able to follow a waveform of typical music, special effects or organ pedal tones with unusually good accuracy and transient response. The brush-commutated, low inertia servo motor generates rotations that are converted to the familiar linear motions in a mechanical fashion. A rotary to linear converter (see illustration) is connected to specially strengthened radiators by means of drive shafts.

Smooth response, very low distortion, and real acoustic power output; we are increasingly of the opinion that no serious playback system should be without these lower two octaves.

I had tended to think of subwoofers as a "Johnny-one-note" used in discos. I simply hadn't realized how vital it is to cover the bottom two octaves smoothly (16-32 and 32-64 Hz).



The June Class

Once in a while, I really blow it. I took a role of film during the class. After the class had left, I opened the camera to remove the film; I found that I had failed to load a roll in the camera!!

Unbelievable. I wanted to call everyone to come back to the class. I felt like the class was incomplete by not having a group picture for my records.

Prosonus™ SRD Studio Reference Disc

Remember the LEDR™ tape that Doug Jones developed to test imaging of monitors and control rooms? Prosonus now markets the tape as a CD. It is now 70 minutes of audio test tones and references. Mary C. Gruszka has listened to it and wrote the following report.

The Prosonus SRD Studio Reference Disc is a compact disc that contains a variety of carefully created test signals on 62 tracks. This in effect gives the user a portable test signal generator that can be used for setting levels, checking polarity, checking coverage of a sound system, etc., etc.

The signals that are included on the disc are the CAVEAT header for audio and videotapes, 17 different sine wave frequency signals at two different levels, two different length sine wave sweeps, TEF™ sweeps for TDS and ETC measurements, full bandwidth white and pink noise, short and long sine wave and pink noise bursts, impulse click, A.S.C. control room monitor test, LEDR test, speaker polarity check, left-right test, pitch references, 88-note piano scale, piano music, and finally a piano chord.

The SRD manual contains a brief description of the test signals as well as some ideas for applications. It notes the names of the creators of the special signals, such as LEDR and TEF, so that the user can write to these people for more detailed information. A bibliography relating to audio, recording, acoustics, and testing is also included.

So what do you do with all of this? The first thing would be to check the CD player you wish to use with the test disc. With the test signals provided, as well as a voltmeter and an oscilloscope (the last two not provided with the disc), you can check the frequency response, amplitude linearity, and probably most importantly, the phase difference between the left and the right channels.

Doug Jones, who was involved in the LEDR test portion of the disc, informed me that he had the disc tested by an independent party who verified that the disc is "interchannel phase accurate." The phase error is less than 0.5% at 16Hz. So any phase errors that are discovered are in the player, not in the disc. Simply put a scope across the outputs and look at the Lissajous pattern. There are a few CD players that look worse than analog tape players in this regard. Phase accurate reproduction of the LEDR test is critical.

Some applications of using the SRD disc as the source for TEF sweeps would be the testing of long lines, satellite uplinks and downlinks, and tape recorders.

This short description of the Studio Reference Disc gives only a few ideas for its use, but I think you can see that it would be a valuable test tool in your shop, lab, or out on the job-site.

TEF™ is a trademark of Crown International

LEDR™ is a trademark of Electroacoustics

Prosonus™ is a trademark of Prosonus

Doug Jone's Address is:

Doug Jones
Electro Acoustic Systems Inc.
1525 Greenleaf
Evanston, IL 60202
312-328-2022

Charles Bilello gave Doug Jones much encouragement to develop the LEDR tape so Doug is contributing $5 from every sale of the disk sold to Syn-Aud-Con grads to the Charles Bilello fund. The list price for the disk is $65. If you send your order to Doug Jones before the end of the year, the cost is $48 plus $2.00 shipping.

Paul Wilbur Klipsch

Paul Wilbur Klipsch at 85 years old rates and receives the world's accolades. He's outlasted his peers and surmounted the phonies. He sits on his own technological and economic summit of success.

Paul is a natural myth maker. When a man of genius chooses to play all of life's roles with zest, vigor and the gifted one's perspective of mankind's folly, the rest of the world tends to regard such genius as "irascible, querulous, a mad genius."

I first met Paul W. Klipsch in 1953. He was in his late forties then and seemed very senior to one in his early twenties. How little I knew then and how much he knew then is still a major humbling influence in my thinking.

Therefore, you can imagine my delight to read in the *American Way* (the airline magazine) issue of March 15, 1989 an article entitled, "A Legend in Sound" by Jim Shahin. The article purports to tell "how eccentric tinkerer Paul Klipsch came to build the world's best-known home stereo speakers - and a $20 million-per-year business - in tiny Hope, AR."

Find a copy if you can and enjoy. Honor "great ones" while they are with us. Their likes may never pass this way again, especially one as original as this Paul Wilbur Klipsch.

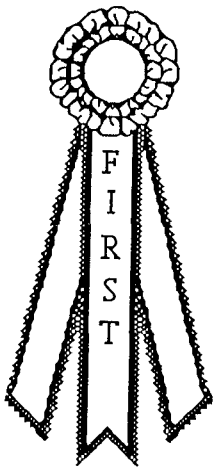
Klipsch and Assoc. Has Been Sold

A couple of weeks after writing the above, a Klipsch dealer, Wayne Staley of Kokomo, IN, told us that Klipsch and Assoc. had been sold to Paul Klipsch's cousin, Fred S. Klipsch.

Fred Klipsch is 48 years old and a Purdue graduate with an impressive resume that reveals business acumen. Fred Klipsch, a resident of Indianapolis, Indiana, purchased Paul's interest in the company, and has assumed Paul's former position as chairman of the board. Paul continues on with the company as a technical and marketing advisor.

We are pleased that Paul Klipsch may now enjoy fully the fruits of his labour as one who has completely deserved his good fortune.

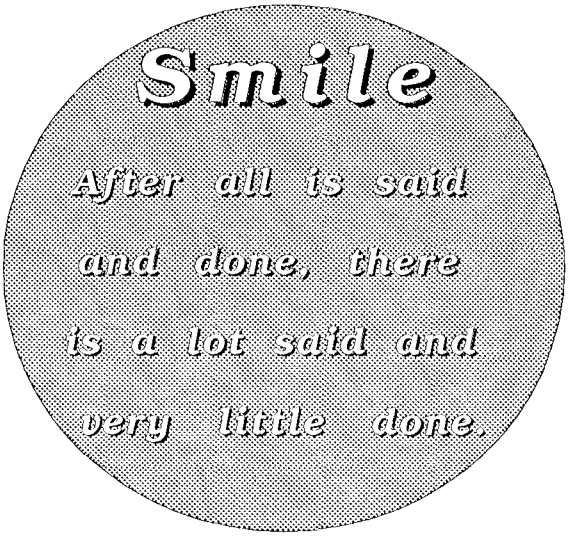
We look forward to meeting Fred Klipsch sometime. And, to Paul we say as Daniel said to King Nebuchadnezzar, "Live forever, oh King."



Biamp Survives!

Biamp should receive the Purple Heart and the Congressional Medal of Honor. They have survived some tough times. We rejoice with them.

Ralph Lockhart, president, and his fine management team were able to make an alliance with Rauland Borg. We don't know all the details yet, but we know both marriage partners well. It bodes well for all concerned: Biamp, Rauland, and their customers.



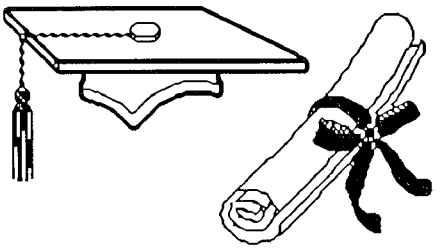
Academic Excellence

We have remarked in the past that we wish we had been fortunate enough to encounter a Dr. Patronis or a Dr. Griener when we were in school. Teachers that we respect are men like the above plus men like Dr. Bertram, Al Grundy, and others who teach with inspiration and enthusiasm.

Others we can now add to the list are Farrel Becker, Adjunct Professor of Physics at American University in Washington, D.C., and Doug Jones, who is the subject of the following news release:

Techron Donates TEF Analyzers to Support Columbia University Sound Engineering Program

Chicago IL, September 22, 1989—To aid in Columbia University's new four-year bachelor of arts program in sound engineering, it was announced today that Techron, a division of Crown International, Elkhart, IN, has made a major donation of six TEF® System 12 PLUS audio analyzers for



student and instructor use. Reflecting a retail value in excess of $70,000, the computer-based units are on loan to the university to provide a unique "hands-on" opportunity for advanced sound engineering students.

Doug Jones, director of the sound engineering program and artist-in-residence at Columbia University, commented on the importance of the acoustic analyzers to the program: "The new B.A. and associates degree programs encompass a broader scope than many similar programs around the country. Rather than developing a course of study that emphasizes only sound recording, we have developed a curriculum which will train professional sound personnel in the varied aspects of sound system design, engineering, testing, and analysis.

"Because the TEF acoustic analyzer is the emerging standard of the industry, it is an excellent tool for students to learn from and become familiar with in the sound engineering program."

Students in the pilot degree program will study sound theory, learn to use TEF software, develop an understanding of acoustic problems and their solutions and ultimately use the TEF system to complete projects such as measuring the performance of microphones, speakers, cross-overs and other components.

We are hearing of more and more university level 2 and 4 year audio programs—many of them under the guidance of Syn-Aud-Con grads. BRAVO!

Spaced Microphones

Why the Häas distance? During playback, signals that shouldn't be heard by the contralateral ear arrive at that ear in the Häas time zone (i.e., around 20 msec.) and the only signal left to be masked is the sounds generated locally by the contralateral sound source.

Many times we read that "spaced" microphones are not considered a good approach to stereophonic recording. Usually the writer has specified neither what he means by "spaced" or by the word stereophonic.

When we talk about "spaced" microphones, the word spaced means that they are the Häas distance apart. This distance can vary according to the liveness or deadness of the acoustic environment but is typically between 15 and 20 feet apart.

Why the Häas distance? During playback, signals that shouldn't be heard by the contralateral ear arrive at that ear in the Häas time zone (i.e., around 20 msec.) and the only signal left to be masked is the sounds generated locally by the contralateral sound source. This means that properly spaced microphone recordings can benefit from the side loudspeaker masking provided by the PAR playback system.

The way that you can tell if you have such a recording is to listen in the conventional manner first with the side loudspeakers turned off and then with them gradually increased in level. When you raise the side levels, if it is a proper recording, the source will jump outside the playback loudspeaker spacing without smearing the image size of solo events.

New TEF Owners

We like to keep a list of TEF owners but we often don't hear where all the analyzers go. We were talking to Peter Miller of Northern Telecom in Calgary. He told us that they had seven TEF analyzers. I don't know how many for sure, but Peavy Electronics in Meridan, MS comes close to seven.

University of Texas at Austin
U.T. Central Receiving
220 Comal St.
Austin, TX 78722

Church of Jesus Christ of
Latter Day Saints
60 North Main—10th Floor
Salt Lake City, UT 84150

Midwest Communications
One Sperti Dr.
Edgewood, KY 41017

Electronic Controls Co.
P.O. Box 7246
Boise, ID 83707

Florida Communications
4407 Vineland #D9
Orlando, FL 32811

American Systems Inc.
820 N. Washington Ave.
Lansing, MI 48906

Audio Technical
1221 Commerce Dr.
Stow, OH 44224

Pyle Industries
501 Center St.
Huntington, IN 46750

McIntosh Laboratory
2 Chambers St.
Binghamton, NY 13903-2699

H P Electronics
9 Newburg Ave.
Baltimore, MD 21228

Cox & Associates
31 Ruscoe Rd.
Wilton, CT 06897

Olson Anderson Co.
106 South McLellan
Bay City, MI 48708

Norcon Electronics
1260 Ralph Ave.
Brooklyn, NY 11236

U.S. Army
Bldg. 3554/89 ME 380
Aberdeen Proving Ground, MD
21005

Commander-Naval Mil. Per.
U.S. Navy Motion Pictures
Flushing & Washington Aves.
Brooklyn, NY 11251

Backstage, Inc.
310 Broad St.
Richmond, VA 23220

Russo Music Center
1989 Arena Dr.
Trenton, NJ 08610

Kimberly Theatrics
98 Line Rd.
Trenton, NJ 08690

Wyle Labs
2001-B Jefferson Davis Hwy.
Suite 701
Arlington, VA 22202

SouthEastern Sound, Inc.
3 McFirrin Ave.
Nashville, TN 37206

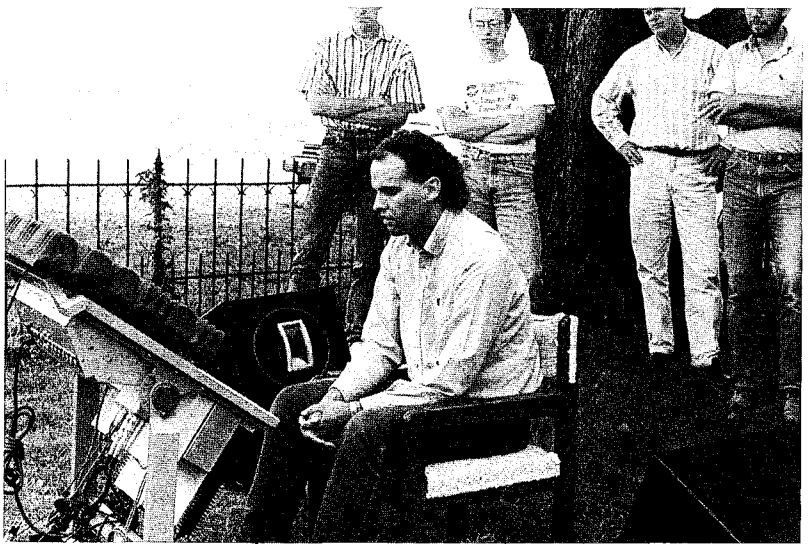
Howell Electronics
2873 Pershing Dr.
El Paso, TX 79903

Industrial Research Products
715 Buss Rd.
Elk Grove Village, IL 60007

Sonics Associates, Inc.
240 Oxmoor Circle
Birmingham, AL 35209

PAR Outdoors

The July class will stand out in our memory as the first time we tried the PAR system outdoors. It's startling, to say the least, to hear all the reflections of a large room as well as the reverberant decay while sitting outdoors.



Professional Services

Acoustical Consultants may list their cards on this page. There is no charge. The only requirements are that you are a full-time consultant, that you have attended a Syn-Aud-Con seminar, and have an active subscription to the Syn-Aud-Con Newsletter. If you would like to be on our Consultants page, send in four (4) business cards for our file.

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RB SYSTEMS *ELECTRONIC SYSTEMS FOR THE PERFORMING ARTS*

ROLLINS "ROLLY" BROOK
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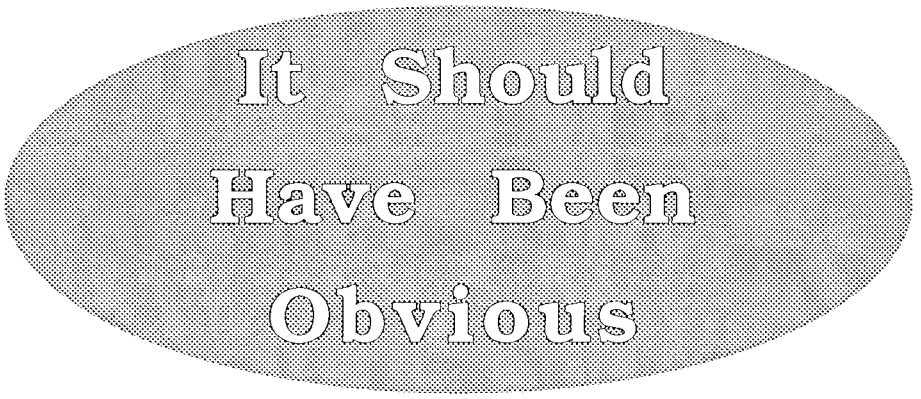


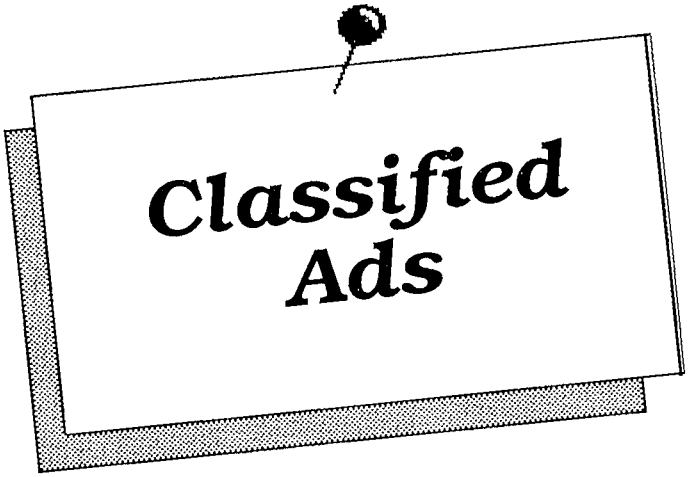
Ted Jones of Indiana University took one look at our latest PAR system with the UREI 809s on the floor and slanted upward to the ears at each side of the head and said, "Do the same thing up front." When we did, we eliminated the need for Sonex behind the listener and had now established the ideal set of angles for interfacing all the loudspeakers to the boundary surfaces of a rectangular listening space.

Next, we put all the control equipment on the floor as well with still another audible improvement. Another fact that should have been obvious but wasn't was our need of a DAT recorder. We persisted for nearly a year trying to make dubs from our older professional Sony cassette recorder. The master tapes were just within acceptability, but the dubs all lost vital imaging clues as well as adding noise that was unacceptable. We hypothesized that it was the inability of the Sony cassette to duplicate the fragile phase relationships above 3000 Hz and out to 10,000 Hz that contain so much of the imaging information.

Going to the Sony TCD-D10 PRO professional DAT recorder solved all of the above problems elegantly. No noise, perfect clones (you couldn't call anything that identical a "dub") and a far superior master because of the remarkably increased dynamic range.

We are now experimenting with new front end IFA-5 preamp mixers from Allen Burdick's "Benchmark Media Systems" that will allow us to mix in a pair of Häas distance spaced microphones along with our ITE microphones. Our hypothesis is that this could possibly help provide more frontal clues to the brain.

We are increasingly coming to the understanding that what ITE recording combined with PAR playback is actually accomplishing is the acoustic encoding—decoding that so many have unsuccessfully sought to do electronically over only two loudspeakers.



FOR SALE:
Techron TEF System 12, dual floppy, version 2.0 software, B&K mic (TEFE-MO1 mic pkg), Epson printer, all manuals and cables. Cost $11,180.00. Sell $8,000.
CONTACT: American Sound & Lighting, 1755 Jenkee Dr., Florissant, MO. 63031 Ph 314-839-3814

FOR SALE:
Techron TEF 12 analyzer. Software included: System 2.0A & EAsyTEF, CPM utilities, AutoTEF, Workbench, 3-D Reverb. All manuals & adapters. $6,000.
B&K type 4007 Microphone kit with phantom supply adapter & 40' extension cable $750.
Crown D-75 amplifier $250.
TOA F-150 monitor speaker w/ tripod $75.
CONTACT: Jerry Pitcher, Charlestown, RI 401-322-1543

FOR SALE:
HP 28S Calculator. Infrared remote printer. Retail $325. Now $175 with extra batteries and paper.
CONTACT: John Odum, 681 Blue Hole Rd, Elkton, KY 42240, PH 502-277-9922

FOR SALE:
Two Altec MR94B horns with throats and Altec 291-16L compression drivers. Excellent condition. Used for testing only $1,000 pr.
CONTACT: Dave Ester (714) 982-5142

FOR SALE:
db Magazines since the first issue - 1966. There may be a few issues missing but the set is nearly complete. We would like someone to have them who would appreciate the collection. $75 includes packing and shipping UPS.
CONTACT: Syn-Aud-Con, 812-995-8212.

FOR SALE:
Pair of Spica Hi Fi loudspeakers. John Bau in Santa Fe, NM makes a beautiful sounding loudspeaker. We bought the pair about 5 years ago. Cost is $200 for the pair + UPS. If interested, we will send a picture.
CONTACT: Syn-Aud-Con 812-995-8212

FOR SALE:
Pair of Ed Long Sonex Model Two MK II TA Hi Fi loudspeakers. Stands 30" high. About 10 years old. Originally over $1,000 for the pair. $200 + freight. Will send picture if interested.
CONTACT: Syn-Aud-Con 812-995-8212

SYN-AUD-CON SPONSORS

Syn-Aud-Con receives tangible support from the audio industry. Seventeen 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 grad 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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