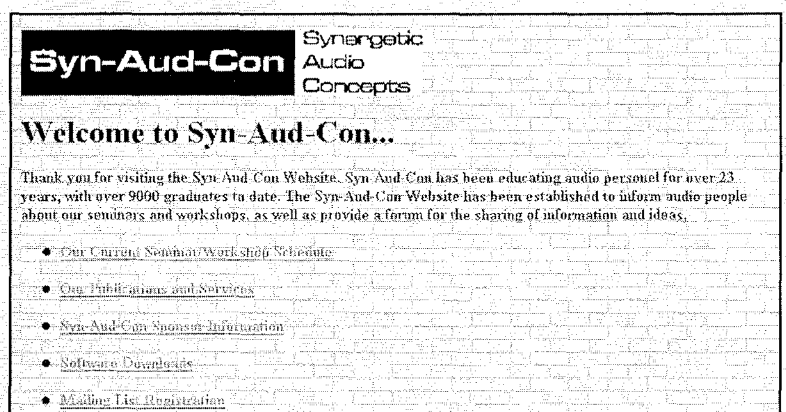
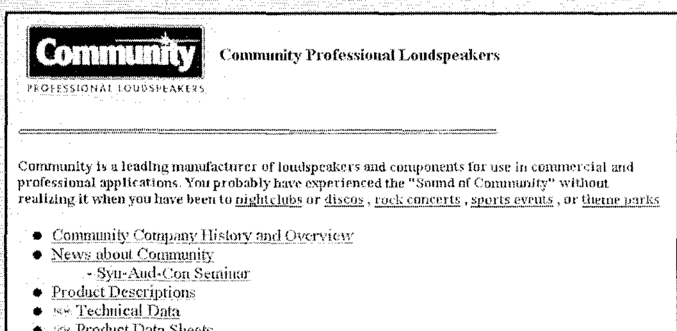
SYNERGETIC SYN AUD CON AUDIO CONCEPTS

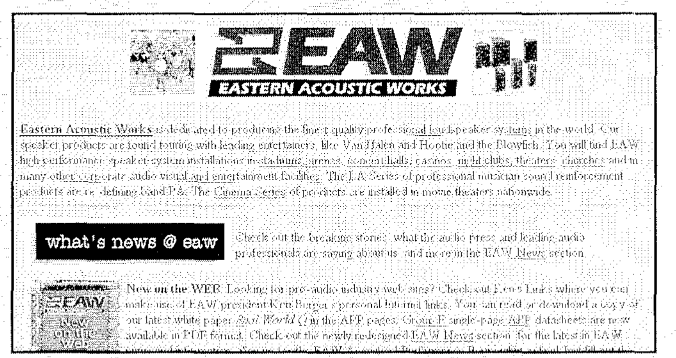
# newsletter

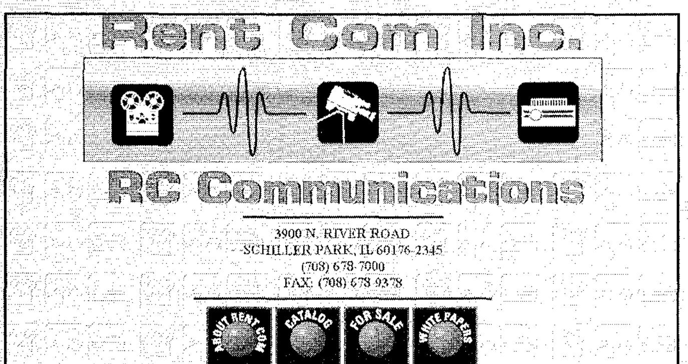
Volume 24, Number 1
Winter 1996


# The World Wide Web




## Audio and the Information Superhighway

SYNERGETIC SYN AUD CON AUDIO CONCEPTS

EXCHANGE OF IDEAS

*I met a man with a dollar*
*We exchanged dollars*
*I still had a dollar*

*I met a man with an idea*
*We exchanged ideas*
*Now we each had two ideas*

Synergetic: Working together; cooperating, cooperative

Synergism: Cooperative action of discrete agencies such that the total effect is greater than the sum of the two effects taken independently.

Editors: Pat Brown, Don Davis, Carolyn Davis,

Design and Layout: Pat Brown, Carolyn Davis



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**Synergetic Audio Concepts**
**8780 Rufing Road**
**Greenville, IN 47124**
**Ph: (812) 923-0174**
**Fax: (812) 923-3610**
**CompuServe 74032,1356**
**Internet patb@synaudcon.com**
**http://synaudcon.com/synaud**

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**Tech Topic "A Project Driven By Speech Intelligibility"**

**When do I renew? - You can check to see when your subscription will expire by checking the mailing label on the envelope in which your newsletter was mailed. In the upper righthand corner beside the name, a date will appear (i.e., 7-94). This means you will receive your last issue with that quarter's mailing unless you renew. Renewal notices will be sent one month prior to your last issue being mailed. You must renew before the next quarter's newsletter is mailed or your subscription will become inactive.**

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# Are You Moving in the 'Net Direction?

Our generation has witnessed more innovations and advancements in technology in its lifetime than just about all of the rest of history combined. It seems that the microscopic switches made possible by the utilization of semiconductors has opened the floodgates to bring the ideas of history to life. Binary arithmetic is nothing new, but now we can do it fast enough and long enough to accomplish some amazing things. When it is all said and done, and history records the advancements made by this generation, the one that will stand out above all others is the communications link known as the Internet. If the Biblical account of the Tower of Babel is literal, and it was the confounding of communication that gave birth to the races and dispersed mankind to the four corners of the earth, it may be the Internet that reunites mankind and restores the communication that was lost. Does this signal the end of an age, or the beginning of one? We will let the scholars have that debate.

The Internet, mostly in the form of the World Wide Web, will change the way that we conduct business in our audio industry. The desktop computer and laser printer have revolutionized the printing and publishing industries. The process of sampling and manipulating images has proven to be superior to analog methods. Video is now experiencing the same revolution, and 1-meter dishes receiving digital signals are fairly commonplace in the landscape. Digital has become the "buzz word" for trying to sell anything electronic. Even our little audio industry is undergoing a revolution, and the principles of digital sampling have become as fundamental as the decibel. With all of this information in digital form, it is now possible to distribute it more effectively than ever before.

As we assemble the material for the Live Sound Workshop manual, class materials from the eight presenters is coming into the office, mostly in the form of overnight packages. A couple of individuals simply e-mailed them to us. They arrived within seconds, and at no cost to them or us. Next year no doubt more will send their material via e-mail, and eventually all will.

How will the Internet affect audio companies? For the sound system contractor, it means being able to get the latest "cut sheet" for inclusion in a proposal without waiting for the mail to come. It will soon no longer be necessary to call the factory and order a dozen sheets for each product in the line, and two dozen of the most popular ones. In fact, it will become unnecessary for the manufacturer to even print such documents in quantity, since they will be available in electronic format at no cost.

I just finished a phone conversation with a customer who was considering taking on a particular product line. Since the company has a Website, I encouraged him to visit it and make use of the information posted there. He did so, and was impressed with the company's offerings. If all manufacturers had Websites, the consultant/contractor could make some fundamental product comparisons without waiting for anyone to mail them a binder.

Need the latest software driver for a computer peripheral? Many are available on the Internet. I was recently pleased to find a ASPI driver for my scanner on Canon's bulletin board, because I needed it <u>now</u>! In fact, we are quickly getting spoiled by the practicality and convenience of this type of support, to the point of getting frustrated with companies that don't offer it. In about every area of business, companies that are not moving toward support of this type are gambling with their future, and ignoring one of the biggest potential money makers available to them. Money makers? More money is made when costs are lowered, and the Internet can cut the cost of providing around- the -clock support to the customer. Most computer software and hardware companies offer support by the following means:

FAX- retrieval system for common problems
Bulletin boards for software drivers, etc.
On-line service forums for questions, etc.
E-mail technical support
Websites for information and drivers
And yes, even technicians on telephones (if you can get through!)

Audio companies would do well to learn from this, and move toward providing electronic support to the customer. Some already do, and many others are working on it.

The Internet is currently a nightmare, full of problems, rough edges, inconsistencies, and worse. But sometimes it works just great, and when this infant grows up (and it is growing up quickly!) it will dominate the transferral of information between humans. And information is what the audio business is. Make a commitment in '96 to educate yourself on this vast resource.

## From Don and Carolyn Davis

# We Created **Synergy** Out of Your **Energy**

We were hard on Pat this Fall. We held him responsible for seeing that NO special activity would be planned during our last class at the farm in October to acknowledge our retirement. Fortunately, they didn't completely follow our wishes. Apparently they included a notice in the Summer Newsletter saying they wanted to put together a book of letters from the Syn-Aud-Con family friends. They prepared a bound book of letters and gave it to us after John Royer presented us the Sagamores award.

We are overwhelmed with the thoughts expressed in the letters by people who attended our seminars and our sponsors. We have read the book many times already and each time we feel that we want to write each person individually and thank them.

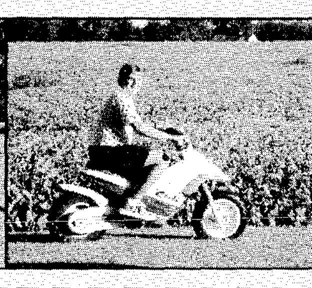
We felt you would allow us to thank you as a group for your wonderful expressions of remembrance and appreciation.

Those who joined with us in class to explore the basics of audio, flavored with the advanced peeks from the likes of Heyser, Patronis, D'Antonio, Peutz and many, many others who gave us of their work, will always be remembered by us. That all of you went on to make Syn-Aud-Con more of a society than a business has left us eternally grateful for your presence in our lives.

Your letters are wonderful. Your participation in our life is memorable. We created Synergy out of your energy. Those with inherent talent sought out Syn-Aud-Con. We were privileged to enable many of you to a more directed use of that talent. Your individual successes were and are our reward.

## Hal Harrison - the future in audio

During our 23 years of teaching seminars we had in our classes several young people in their teens but mostly late teens. When I read Hal Harrison's registration information sheet prior to the September class and saw that he was 14 years old, I called Pat Brown to be sure that he felt that Hal would get his money's worth - as everyone who has attended a Syn-Aud-Con class at the farm knows, it requires a sizable investment. Pat told me that he had talked with Hal's parents in South Carolina and that they were very supportive of Hal's attending with his friend, Mike Stokes. (Fourteen is the youngest of anyone attending our classes. Pye Clark, 14, attended our Syracuse class in the late 1970s. Pye, as her name implies, is very good at math.)



Hal was a joy to have. He was enthusiastic about the class, the animals on the farm and Carolyn's motorcycle. Hal loves computers, computer programming and audio. Perfect for his future and the future of audio. As Don and I end our career in audio, it was rewarding to launch a new career in audio. *cpd*

# A Christmas and New Year's Message to All Our Audio Friends

**1995** - the year we started a new career: some call it retirement, we call it a shift of priorities, an active involvement in our farm and community. During the 23 years of wonderful and total commitment to Syn-Aud-Con, our life and interests were "on the road." Today it is never-ending projects at the farm, and with friends and family. An expert on the subject, Ken Wahrenbrock, told us that if we weren't busier after retirement than before, then we were not truly retired. We are truly retired! Pat and Brenda Brown have carried Syn-Aud-Con off to the Louisville area and are doing a great job of securing its future possibilities.

Soon after starting Syn-Aud-Con in the early 70s, we sold our Porsche and gave up driving high performance cars; now Don has gone to the Boundarant driving school to refresh his skills and is pleased to be in a real driving car again.

After years of spurning television, we have DSS now and we are C-SPAN junkies - though we still feel that watching television during daylight hours is form of decadence. We are on Internet now, spending about an hour a day exploring the Wonders of Web. Carolyn has read 10 books since turning Syn-Aud-Con over to Pat & Brenda. Don never let anything interfere with his constant reading schedule of dozens of books each year.

The "big thrill" this year happened in May: we had the privilege of hosting Col and Mrs. Jeff Cooper for the Indy 500 and we had a full siren and lights police escort from our hotel to and from the track tower, courtesy of a Gunsite grad from the Indianapolis Police Department - his way of showing respect for Mr. & Mrs. Cooper.

Pete, Callie, and Rascal continue to provide absolutely unique feline contributions to our sense of humor. We are now body-guarded on our walks in the woods by five dogs - Patch, Pedro, Roe, Versus, and Wade. We have dozed new hiking, motorcycling and wood gathering trails through our woods - the farm stretches 1 mile along a north and south line.

The horses, Jesse and Red, are older and don't often enjoy a lope across the pastures now, but they do love to be brushed and groomed. Poncho, the llama, continues to "kiss" all the visitors. Bo and TuTu, the goats now are confused as whether they are llamas or horses.

Much to our surprise, winter has become one of our favorite seasons at the farm. A wood stove, a good book, a roast in the oven, snow on the ground, and no pressure to go anywhere spells heaven on earth. We get plenty of exercise doing the farm chores, caring for a big garden, swimming, hiking, motorcycling, target shooting — and even skiing. We have an outstanding facility with snow making capability less than an hour away with 15 runs - the only problem is that the longest run is less than 1/2 mile, but it is still fun.

The books, music, and wonderful panoply of nature around us all vector towards a greater sensitivity to that silent inner voice that comes when the senses are silent. Both of us are enjoying robust health, harmony and happiness. Once again we recognize and acknowledge that God's blessings have overwhelmed us and we are grateful listeners.

*cpd & dbd*

## Sagamore of the Wabash

At the conclusion of the last class held at "the farm", John Royer made a presentation of two "Sagamore of the Wabash" awards.

The Sagamore of the Wabash is "the highest honor which the Governor of Indiana bestows." Needless to say, Don and Carolyn were deeply touched that John had personally gone to the Governor and suggested the award. To be a "prophet with honor" in our home state provides a warmth of heart only friends such as John are capable of thinking of and acting upon.

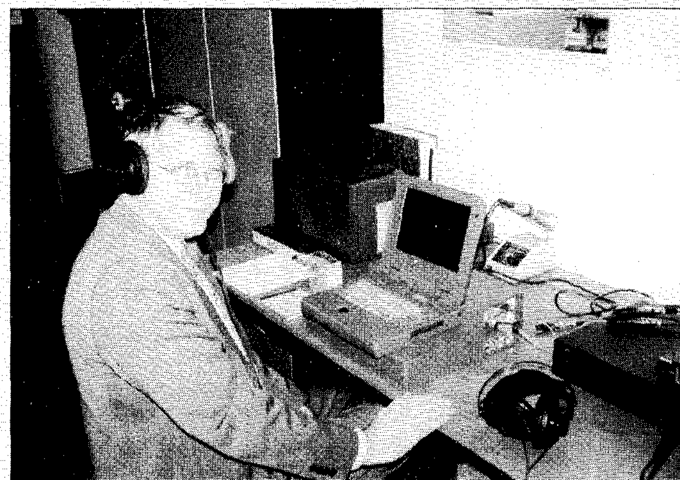
*cpd & dbd*

Acoustics Easy and Fast.

# Mark IV Joins the Growing List of EASE/EARS Users

Dr. W. Ahnert of Acoustic Design Ahnert (ADA) of Berlin, Germany is proud to announce the addition of Mark IV Audio to the growing list of EASE/EARS users. Under this agreement, Mark IV will distribute EASE to its large customer base. Renkus-Heinz continues to distribute the programs worldwide to all other interested parties. The EASE program will replace AcoustaCADD as Mark IV's premiere design program. The company will continue to support and develop its own loudspeaker selection and aiming program, AcoustaQWIK. EASE/EARS has been updated to directly import AcoustaCADD files. The program has already experienced the influence of Mark Ureda, whose array modeling software wowed the attendees of the Horns II workshop in Columbus, OH last summer. A Cluster Lobe module will be added to a future release of EASE, to allow complex loudspeaker arrays to be modeled at the drawing board stage of a project. A Windows version is also in the works.

We watch with enthusiasm as this product continues to evolve. pb



*Dr. Ahnert has been hard at work updating his EASE program to work with AcoustaCADD files. While visiting him in Berlin, I was treated to a demonstration of the "head turning" algorithm for room auralization.*

# S. N. Shure Dies at 93

*S. N. Shure*

Sidney N. Shure, and industry leader in audio electronics for over 70 years, died on October 17 in Chicago. His privately-held company, Shure Brothers Incorporated, is the world's largest manufacturer of microphones.

Known professionally as S.N. Shure, he was born in Chicago in 1902. The son of a merchant, Mr. Shure learned the facts of business life at an early age. He attended Austin High School in Chicago and took his B. S. degree in Geography at the University of Chicago. S. N. Shure was 93. He is survived by his wife, Rose Shure; his son Bob Shure; and his daughter Myrna Shure.

Syn-Aud-Con owe's much of its success to people like Mr. Shure, who has been a longtime supporter. We mourn the passing of such individuals and hold them in high regard for their contributions to the industry and humanity.

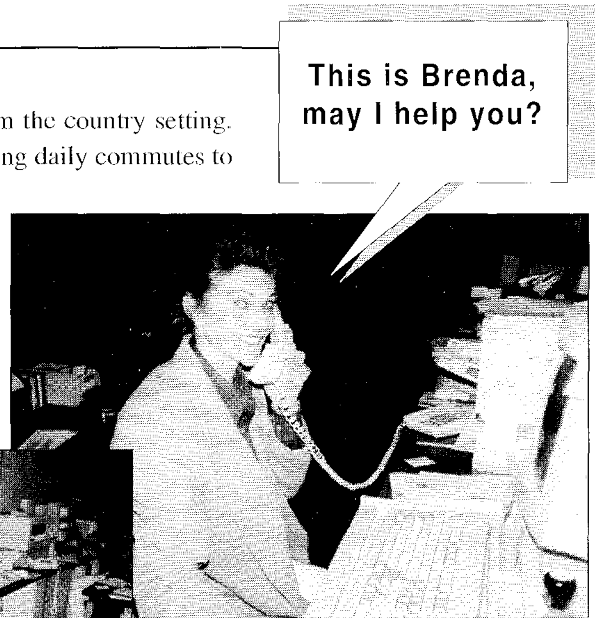
# The Greenville Office

The Syn-Aud-Con office just can't seem to get away from the country setting. The office has moved from one rural location to another, making daily commutes to the city unnecessary. Greenville is a small town about 20 minutes from Louisville, KY. We have all of the essentials; a bank, post office, pizza parlor and video store.

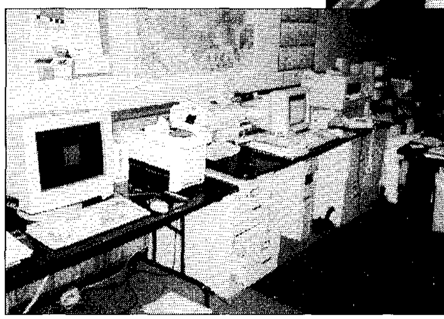
The move to the Greenville office is complete, having begun in July. The facilities include the office (shown below) which is run (ruled) by Brenda, and "Dad's office," where Pat must go to get anything done.

The Greenville office is just a few scenic miles from the Joe Huber Family Farm in Starlight, a popular tourist site where our Southern Indiana classes will be held in 1996.

pb

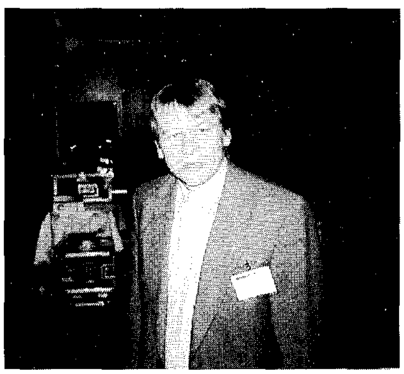



*That pleasant voice that answers the phone is Brenda Brown.*



*Computers abound, but we need more!*

# Audio Education in Europe

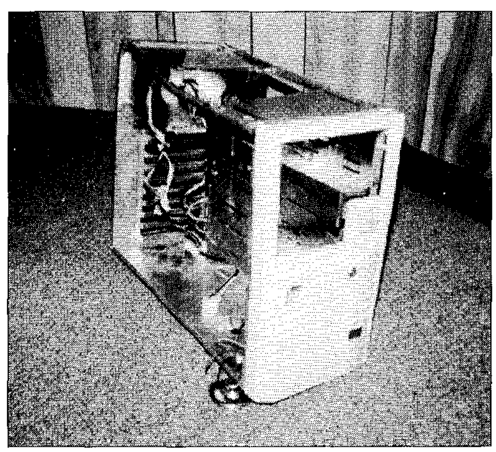


*Volker Loewer*

One of the presenters at the Avitech Seminar in Frankfurt, Germany was Volker Loewer of IFB, a well-known consultant in Europe. Mr. Loewer also teaches training seminars in audio and acoustics. It was fascinating to listen to him as he described the same audio and acoustic principles that we teach in our classes, but from a German perspective. IFB was also represented by Ms. Sabine Heymann, who gave several interesting lectures on sound system basics. We look forward to working with Volker Loewer, Sabine Heymann and Dieter Michel of Prosound magazine on future educational projects. The Internet has certainly broken down the distance barrier, since we can now exchange information with Germany as though it were just down the hall. pb

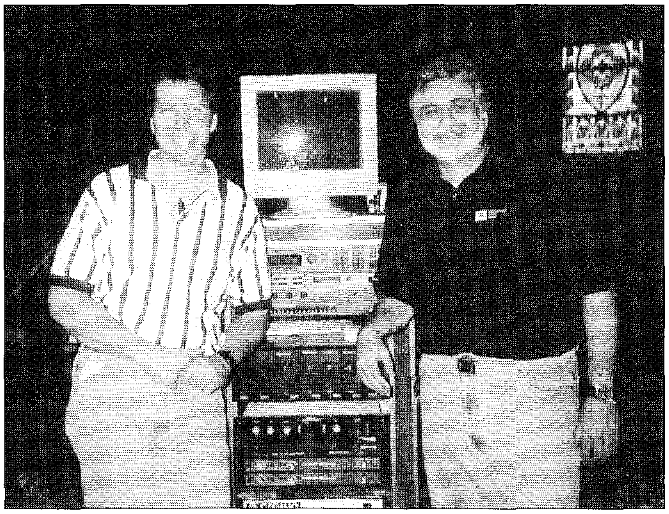
# Pentium Problems

Having your computer bounce down an escalator is no way to start a morning in Frankfurt, Germany. Pictured to the right are some of the remains (it was in a carry-on case). Fortunately, I had my "analog" presentation (printed overheads) with me to complete the class sessions for Avitech and Mark IV. Good insurance and complete backups allow for quick recovery from such misfortunes. On the return trip, Frankfurt airport security was unsettled because I couldn't turn it on to prove that it wasn't an explosive device. pb



# Fall River Class Sets Precedent for Future Sound Engineering Seminars

Paul Hanoud, our Northeast Syn-Aud-Con representative attended the Secaucus, NJ seminar in the fall of '94 and went away excited. During the course of the class he conceived the idea of a class at "The Central Complex," a historical and architecturally interesting site in Fall River, MA. The "Central" is an old Cathedral that was saved from the wrecking ball by a group of local activists. It has become a cultural island for Fall River, and was the site of an Aerosmith music video last year. With the help of Paul and his wife, Nancy, the necessary arrangements were made to hold a seminar at The Central.



*Pat Brown and Paul Hanoud*

This seminar was quite an experiment for Syn-Aud-Con. The decision to leave the security and predictability of the hotel environment and hold a class "in the field" brought a myriad of logistical problems with it. The "behind the scenes" work that Brenda must do to arrange a seminar increased about tenfold, as every detail became a variable.

We had several things that we wished to accomplish at Fall River. First, we wanted an acoustical environment that would allow some "real world" measurements to be made. The 350,000 $ft^3$+ cathedral did nicely for that. The concepts of reverberation, critical distance and directivity factor "come alive" when the explanation can include a demonstration. A class project included the estimation of the octave-band $RT_{60}$'s using our test CD, and then verifying the estimates with TEF measurements. The correlation was excellent, proving what we have been saying all along; your ears are the best instrument that you possess. The next step was to measure the critical distance, using only a sound level meter, of a device with a know directivity factor. Our Community M200/PC494 horn and driver did nicely. We then substituted "unknown" loudspeakers and derived their directivity factors by measuring how far critical distance moved. Afterward, many told us that this session alone was worth the price of admission, since the role of "Q" was now permanently engraved in their minds.

Day two was gain structure day, as we worked with levels from talker to listener using the Syn-Aud-Con slide rule. The last session of the day was spent in smaller groups calibrating real systems, which included equipment from an array of manufacturers. Evenings provided time for questions and answers, and the sharing of individual experiences from attendees.

The last day of the seminar involved the setup and calibration of several types of loudspeaker arrays, which included single source, left-center-right, and a JBL box array system. There was plenty of time for "hands on" TEF and real-time analyzer measurements.

Just like audio systems, there are no perfect seminars. But what made Fall River a successful experience for attendees was the "synergy" that occurs when a bunch of audio folks get together in a big room with lots of audio equipment. It was an important milestone in the evolution of our seminars, and many of the things that worked at Fall River will be used in the future. We are grateful to all who agreed to participate, which included a capacity crowd from the East coast and beyond.

pb

## FALL RIVER SCENES

# New Frontier Electronics

We are always glad to see innovative individuals bring new products into the audio marketplace. Virtually every major manufacturer today can be traced back to an entrepreneurially minded individual that saw a hole in the marketplace and moved to fill it. Our Fall River class had two such individuals in the person of Michael McCook and Tony Keane. Their company, New Frontier Electronics, manufactures an FFT real-time analyzer, surge protection systems, and an A/V signal tracer. Michael attended our farm class in Indiana several years ago. In addition to manufacturing, Michael is a sound contractor, specializing in cruise ships.

We used the RTA 2010-EX (the companies' single rack space dual-channel FFT analyzer) to monitor the noise levels at the Fall River seminar site, and as an oscillosope for doing gain structure. The long-term SPL mode revealed the noise levels present during the night time hours when no one was in the facility, revealing that the small town "comes to life" at about 4:30AM. pb

*Michael McCook and Tony Keane*

# Three Levels of Syn-Aud-Con Training for 1996

1996 will mark the introduction of two new training seminars from Syn-Aud-Con. Our Sound System Design Seminar, a mainstay in the audio industry for 23 years will continue, and will be supplemented by two more levels of training. Here is an overview of what one can expect at each seminar.

**Sound System Operation Seminar**

This seminar is designed to meet the needs of the individual whose responsibilities are mainly sound system setup and operation. Many individuals do not need an in-depth understanding of the engineering principles of sound reinforcement to do their jobs. This seminar focuses on the practical aspect of sound reinforcement, from microphone selection and placement, to maximizing the acoustic gain of the system. At the end of the two day program, the individual will be able to :

*Choose the proper microphone based upon sensitivity and coverage*
*Determine the best mic placement for various sources*
*Understand the functions of each part of a mixing console*
*Do a "quick calibration" of the signal processing chain to optimize headroom and signal-to-noise ratio*
*Place loudspeakers for best coverage and performance*
*Properly calibrate a graphic equalizer*
*Maximize the system for acoustic gain*
*Perform basic troubleshooting tasks*
*Check system wiring from mic to loudspeaker*

This "math free" seminar will play an important role in introducing individuals to sound system operation, and provide a foundation for further Syn-Aud-Con training. All concepts will be applied in a very practical and "hands on" manner, allowing the individual to immediately implement them in the operation and maintenance of sound systems.

**Sound System Design Seminar**

If your duties include the actual design of the sound system, the Sound System Design Seminar will provide the tools needed to complete this task. Some of the topics presented include:

*The decibel in the acoustical and electrical domains*
*Designing for the human auditory system*
*Room acoustics and evaluating the acoustic space*
*Estimating reverberation times*
*Predicting Speech Intelligibility*
*Finding and using critical distance*
*Measuring parameters of signal processing devices*
*Calibrating and documenting the gain structure of the signal processing chain*
*Using the decibel from talker to listener*
*Measuring and predicting acoustic gain*
*Loudspeaker arraying techniques*
*Proper use of microsecond delay devices*
*Equalization techniques*
*System measurements and instrumentation*

**Advanced Sound System Design Seminar**

For those who have attended our Sound System Design Seminar, or already have proficiency in the concepts that it teaches, we offer an advanced training seminar. Modern technology has given us many tools for designing and implementing systems, and the focus of this seminar is the integration of those tools in the design and implementation of sound reinforcement systems. The seminar is designed to allow the professional or serious amateur to stay abreast of the latest developments in our industry. Some topics that will be covered include:

*Modeling rooms with the computer*
*Modeling arrays with the computer*
*Auralizing rooms at the drawing board stage*
*FFT and TDS measurement techniques*
*Measuring speech intelligibility*
*Loudspeaker measurements*
*Electronic enhancements to room acoustics*
*Intelligent sound systems*
*Remote amplifier control*
*Digital Signal Processing*
*Computer-control of complex systems*

Syn-Aud-Con sponsors will play an important role in providing specialized training on specific subjects.

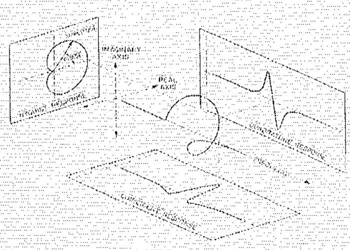
We feel that these three levels of training will provide individuals and companies with skills that can be applied in any area of sound reinforcement, as well as a means for progressive education in the audio field. The current schedule for these classes can be found on the back page of this newsletter, and on our World Wide Website. pb

## Sound System Operation Seminar Overview

**Day One**
Basic Theory
System Components
- Microphones
- Mixers
- Equalizers
- Processors
- Amplifiers
- Loudspeakers

Interconnecting Components
Setting Levels
Testing Polarity
Basic Grounding Practices
Basic Wiring Practices

**Day Two**
System Setup and Operation
Loudspeaker Basics
Loudspeaker Placement
Microphone Placement
Mixing Techniques
Adjusting the Equalizer
Reducing Feedback



## Sound System Design Seminar Overview

**Day One**
Design Prerequisites
The Human Auditory System
The Decibel in Acoustics
Room Acoustics
Basic Acoustic Measurements
Speech Intelligibility

**Day Two**
System Gain Structure
The Decibel in Electronics
Ohm's Law
Component Specifications
Levels and Impedance
Grounding and Shielding
From Talker to Listener Using the Decibel

**Day Three**
Loudspeaker Arrays
Acoustic Gain
Loudspeaker Parameters
Arraying Techniques
Calculating Coverage
Array Synchronization
Equalization Techniques

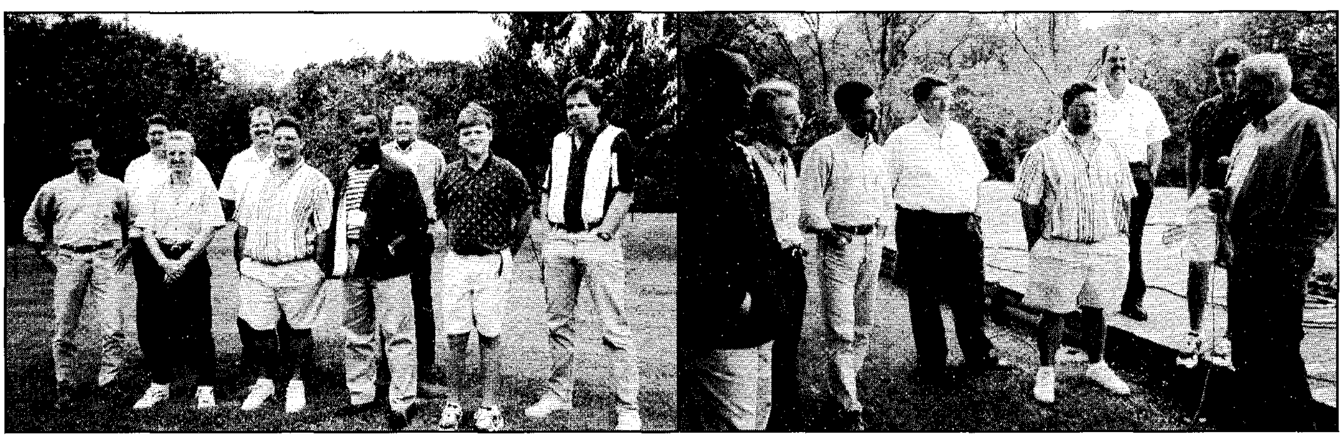
## Advanced Sound System Design Seminar Overview

**Day One**
Computer Design Methods
Room Modeling
Ray Tracing Methods
Image-Source Methods
Array Modeling
Auralization Techniques

**Day Two**
Advanced System Measurements
The Impulse Response
FFT Measurements
TDS Measurements
The Analytic Signal
The Heyser Spiral
Measuring Speech Intelligibility
Noise Measurements
Array Measurement and Calibration Techniques

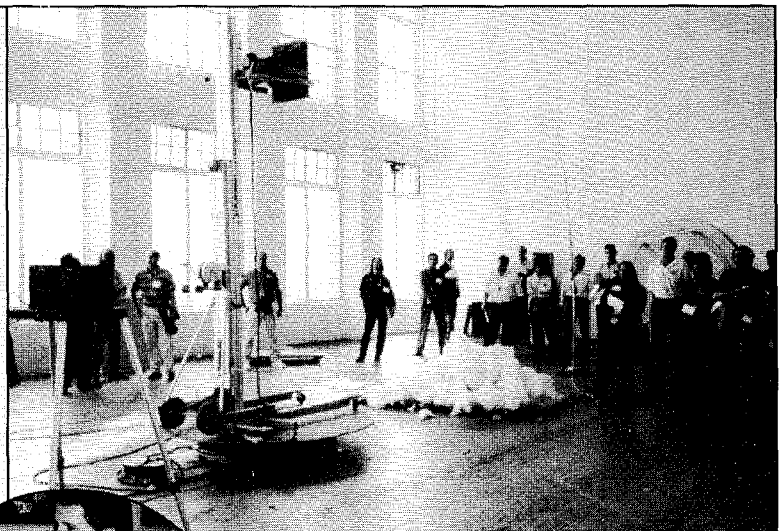
**Day Three**
Computer-Controlled Systems
Electronic Acoustic Enhancement
Intelligent Sound Systems
Multiple Amplifier Control
Digital Signal Processors
Advanced Equalization Methods
Automatic Mixers
Computer-Controlled Signal Routing

## *September Class at the Farm in Indiana*



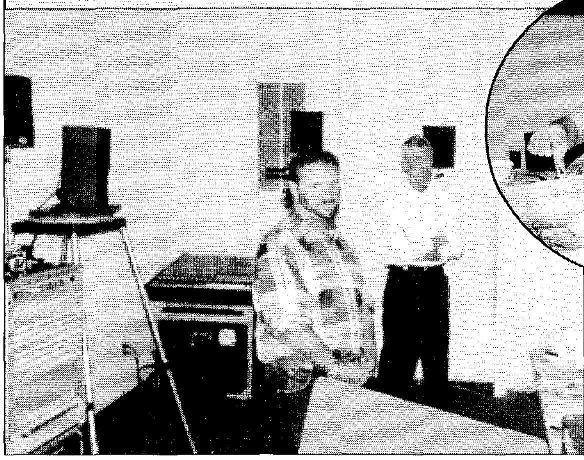
# Eastern Acoustic Works Hosts Syn-Aud-Con Seminar

EAW hosted a three-day training seminar for its international customers prior to the AES show. In addition to the classroom sessions, attendees enjoyed a factory tour, a New England clam bake, and the hospitality of the EAW staff. We hope to return to Whitinsville next spring for another class. pb



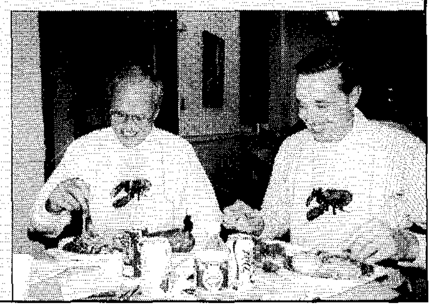
▲ *"The Pit" provides a place for acoustic demos.*

*Ken Berger and Frank Loyko address the class.* ▼



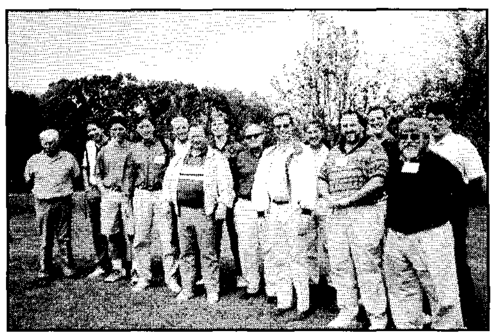
▲ *Is this how you do it? A first try at clams in the shell.*

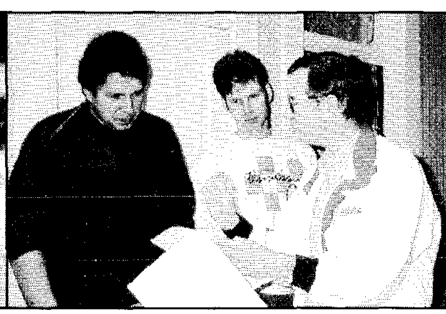
*Larry Walburg and Larry Brandsen enjoy an evening lobster feast.* ▼

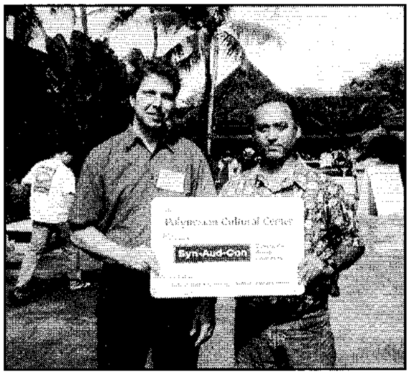


# October at the Farm

The October Farm Class marked the end of an era of classes at the Farm. Don and Carolyn's retirement from audio has ended the era of classes and workshops held in the 100+ year-old house. We all hold fond memories of those classes, and I count myself fortunate to have participated in them for the past three summers. Those who made the pilgrimage to a Farm class hold a piece of audio history that they can cherish for a lifetime. Far from dormant, the Farm continues as Don and Carolyn's place of experiencing all that life in southern Indiana has to offer. pb





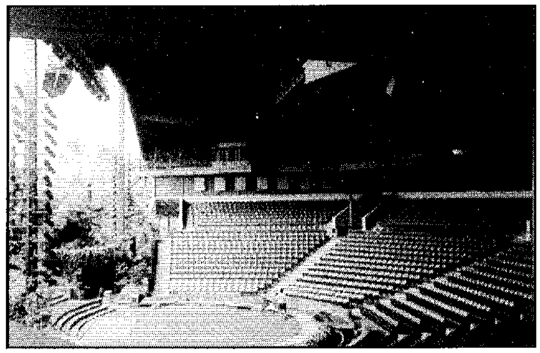



*Special thanks to Stephen Crowell and the staff of the Polynesian Cultural Center*

# A Seminar in Paradise

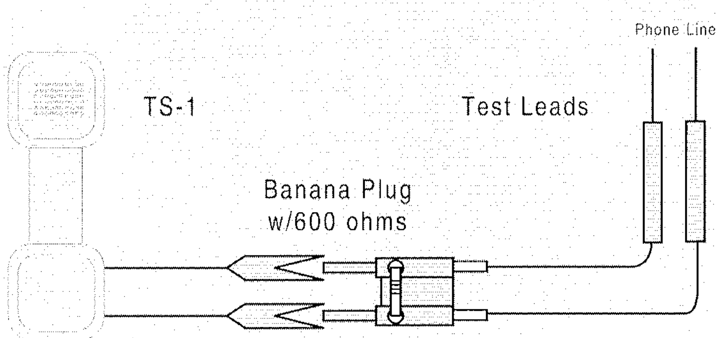
The Polynesian Cultural Center is one of the finest attraction of the Hawaiian Island chain. The Center preserves and presents a cultural overview of many of the peoples of the South Pacific. We were honored to be invited to present a class for the Center's staff, as well as anyone else that wished to attend. The class was conducted at Brigham Young University's Hawaii campus, and stretched over five days to allow the Center's staff to fulfill their mixing and maintenance duties. The Center maintains several large systems, including an Imax theater.

Unfortunately, an immediate return flight was unavailable, so Brenda, Ashley (oldest daughter) and I were "forced" to stay an extra week.    pb



# Troubleshooting Phone Lines with the TS-1

The TS-1 Test Set from Music Supply and J.W. Davis has found yet another application around the Syn-Aud-Con office. A problem had developed with one of the incoming telephone lines, so some troubleshooting became necessary. The impedance of the TS-1 is about 2000 ohms, which is not enough to signal an "off hook" condition to the phone company when the unit is placed across the line. A banana plug and 600 ohm resistor (see diagram) allows one to listen for a dial tone at various telephone jacks. In our case, a quick check at the main entry revealed that the line was okay, and the problem was traced to a telephone that had become defective (no doubt the kids did it!) and was loading down the line. The banana plug/resistor "accessory" had found a permanent place in my TS-1 storage pouch.    pb



# Anatomy of the Decibel

The human auditory system has an incredible dynamic range. The eardrum is sensitive to changes in pressure, and the smallest pressure change that it can detect is about 20 microPascals. Of course, it also has a limit of what it can withstand without damage, which is about 200 Pascals. The difference between these two extremes is:

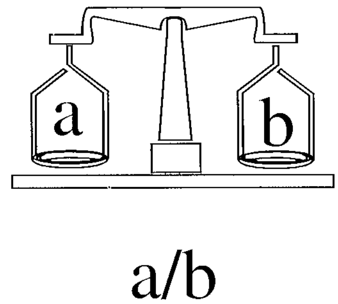
$$\text{Dynamic Range} = \frac{200}{20(10^{-6})} = 10{,}000{,}000$$

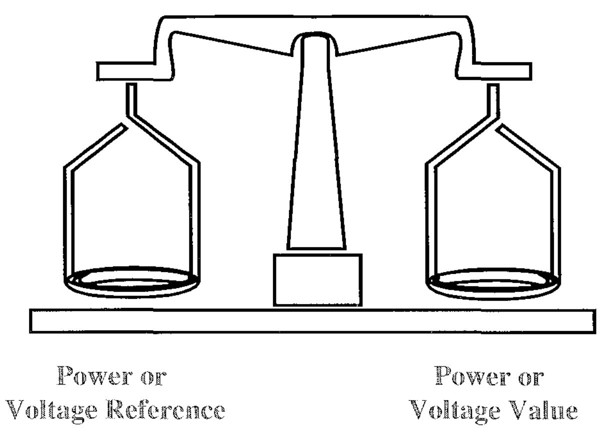
Such a large dynamic range can be tedious to deal with mathematically, because the powers, currents, and voltages in sound systems extend over similar ranges.

Since no one likes to work with extremely large numbers, a mathematical method was derived to reduce these numbers to a more manageable level. This method makes life much easier for audio personnel, but as with any such tool, time must be taken to become familiar with it. In this article, we shall dissect the decibel to gain insight into what it does and why we need it.

**Compare**

The first step in changing a number to decibels is to do a comparison. This is actually the way that our auditory system works. We live at the bottom of a sea of air, and the pressure on our bodies from this air is about 1000 lbs/ft$^2$. We don't "implode" because we are equalized. This pressure is a sort of "acoustic bias" for the eardrum, and every change in pressure is relative to this absolute pressure. In short, when we hear, our auditory system has detected a ripple in the static atmospheric pressure. This can be visualized by considering an old fashioned scale. A known quantity is placed on one side, and an unknown on the other. A calibrated pointer indicates the difference between the two. For sound levels, the quantities being compared are sound pressures, and there are two ways that this can be done. The first is to compare all unknown pressures against a reference pressure. Since our hearing threshold is 20 microPascals, this is the value normally used. All such measurements are termed "absolute" because they are comparisons made with a common and precise reference. Another useful method is to compare two unknown pressures with each other, and not a common reference. A comparison of this type always yields the difference between the two, which many times is all that we need to know. Measurements made in this manner are called "relative" measurements, because they reveal how the quantities are related to each other. When engineering sound systems, there are a number of quantities that we are forced to work with, and each has it's own standard reference that is used for absolute measurements. They are as follows:




| Quantity | Reference |
|---|---|
| Sound Power | $10^{-12}$ Watt |
| Sound Intensity | $10^{-12}$ W/m$^2$ |
| Sound Pressure | .00002 Pascal |
| Electrical Power | .001 Watt |
| Electrical Voltage | 1 Volt or .775 Volts |

Each of these "power-like" quantities is important at various stages of the sound system, and each is useful in it's own way and for it's own purpose. When an unknown "power-like" quantity is compared with one of these absolute references, the result is a ratio that describes the relationship between the two quantities.

**Compress**

The second step in making these large numbers more manageable is to apply a principle from everyday life. When the suitcase is too full, some compression aids us in closing it. When the trash can is too full, some compression can delay a trip to the garage. The second step in working with large numbers is to apply a mathematical compressor. A simple and straightforward tool is the logarithm. This is the expression of a number as a power of 10. This allows us to express the numbers between 0 and 1,000,000 as a number between 0 and 6. That's a lot of compression! If a comparison between two power-like quantities yielded a ratio of 1000 (not uncommon), taking the log of this ratio would yield the number 3. If the ratio were 10,000 it would yield the number 4, and so on. Some common ratios and their logarithms are shown below.

| exponential form | ratio | log |
|---|---|---|
| $10^0$ | 1 | 0 |
| $10^1$ | 10 | 1 |
| $10^2$ | 100 | 2 |
| $10^3$ | 1000 | 3 |
| $10^4$ | 10,000 | 4 |
| $10^5$ | 100,000 | 5 |
| $10^6$ | 1,000,000 | 6 |

Another useful by-product of converting numbers to powers of 10 is that addition can be substituted for multiplication, and subtraction for division, making these two mathematical operations the most widely used in sound system design. Using logarithms, the multiplication of two quantities can be replaced with the addition of their logarithms; a neat trick for the sound system engineer.

The unit for a log of a ratio between two power-like quantities is the Bel, in honor of the inventor of the telephone.

Logarithms make life easier for all, but there is one additional step necessary to complete the conversion of a number into decibels.

**Scale**

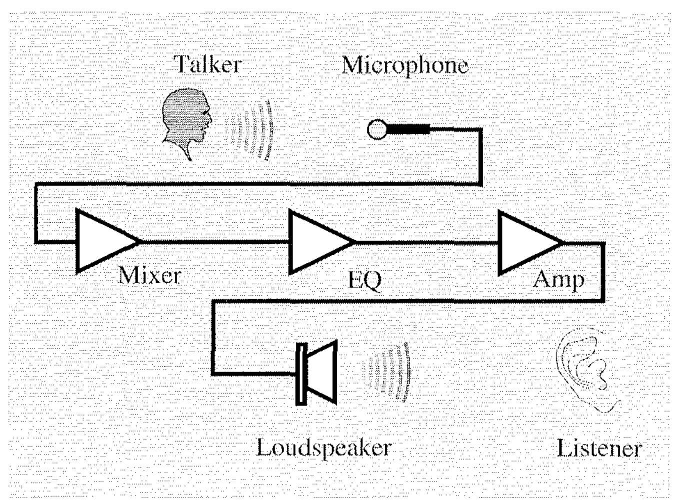
In the course of applying the mathematical compressor, the number range required to describe the perception of the human auditory system (and the workings of a sound system) was reduced considerably. In fact, the number range has become so restricted that it would be difficult to be precise without using the decimal point and an additional number. For this reason, the final step of the conversion is to scale the result of the logarithm step by a factor of ten. With this step, the numbers between 0 and 6 from the previous example are expressed as numbers between 0 and 60. This makes the decimal point unnecessary in all but the most precise situations.

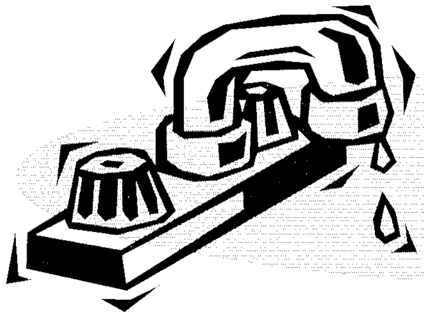
| ratio | Bels | decibels |
|---|---|---|
| $10^0=1$ | 0 | **0** |
| $10^1=10$ | 1 | **10** |
| $10^2=100$ | 2 | **20** |
| $10^3=1000$ | 3 | **30** |
| $10^4=10,000$ | 4 | **40** |
| $10^5=100,000$ | 5 | **50** |
| $10^6=1,000,000$ | 6 | **60** |

The decibel has earned a permanent place in the vocabulary of the audio professional, and mastery of this important concept is simply understanding what it does and why we need it. In future articles, we will be applying the decibel to various parts of the sound system, beginning with the talker and ending up at the listener. The anatomy of the decibel can be described in three words:

Compare ⇨ Compress ⇨ Scale

pb





# Hot or Cold?

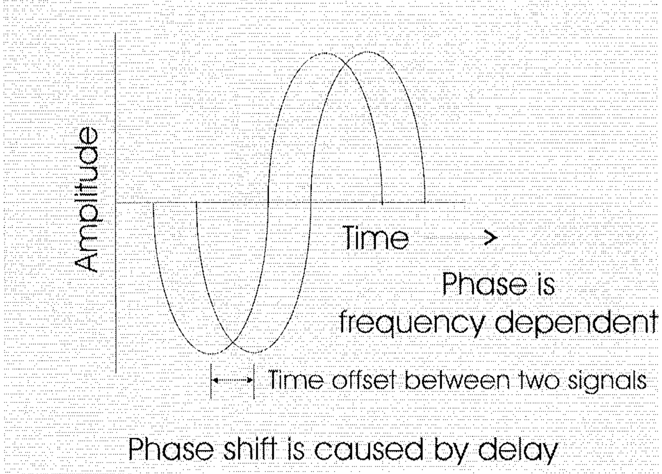
*Polarity testing in sound systems to identify and prevent problems*

The relationship between polarity and phase is not always clearly understood. Sometimes the two terms are erroneously used interchangeably. Phase and polarity relationships both occur in the electrical environment and the acoustical environment. While the phase of a signal varies with distance and frequency, the device's polarity remains constant. That is, reversing the polarity of a speaker will change the phase of every frequency by 180 degrees.

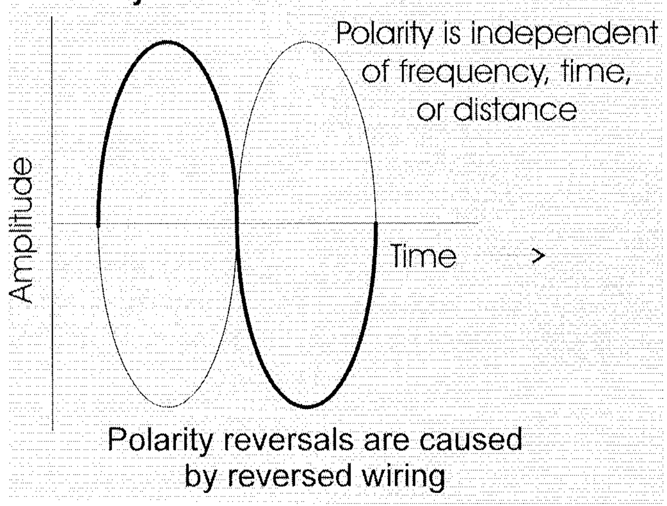
**Polarity**

Polarity can be defined as: Given an overpressure (compression of air molecules) on the diaphragm of a microphone, then there will be a positive (+) electrical voltage at PIN 2 (if the mic output is an XLR connector) or at the TIP of a 1/4" (TRS or 1/4" unbalanced) connector. A positive voltage applied to the + terminal of a loudspeaker produces an overpressure in the atmosphere (the speaker cone moves forward). The converse is true: a negative (-) voltage at the (+) terminal of the speaker produces rarefaction (increase of distance between air molecules) as the speaker cone moves back. A rarefaction at the microphone diaphragm produces (-) voltage at the microphone output. Polarity has only two states IN (0 degrees) or OUT (180 degrees). For sound system purposes, polarity is absolute.

**Phase Shift Illustrated**




**Polarity Inversion Illustrated**




**Phase**

Phase is relative, varying from 0 to 360 degrees where 360 degrees represents returning to 0. For a given frequency, phase is relative to time.

For example: A 56 Hz tone has a wavelength of approximately 20 feet, i.e., standing 20 feet from the speaker producing the 56 Hz tone, the phase at your location at 17.8 ms (one period of 56 Hz) would be 0 or 360 deg. At 10 feet (at 17.8 ms) it would be 180 deg.

If we double the frequency to 112 Hz, the wavelength will be 10 feet. If we stand at 10 feet (and wait 9 ms), we have 360 degrees of absolute phase shift, measured from the time the sound left the loudspeaker. We have zero degrees phase shift *relative* to where the wave is in it's cycle. At 20 feet (and 18 ms) we have 720 degrees of absolute phase shift, which also puts us back at zero for relative phase. For both tones, at the loudspeaker, we have zero feet distance, zero time and zero degrees phase shift. In both cases, and at all distances, polarity has only two states; IN (0 deg.) and OUT (180 deg.)

When two sources emit the same tone (frequency), be it electrical or acoustical, and if the tones are combined at a given point, if they are out of polarity, they will cancel each other. If we *delay* one tone in respect to the other, there will be summation or cancellation, depending upon the actual time that one signal is delayed. In sound systems we often have only partial control over phase. What we do have, is absolute control over polarity.

In even the simplest sound system there are numerous opportunities for polarity to become reversed. The more complex the system, the more chances for out-of-polarity conditions to occur. Our responsibility, as creators of these systems, is to eliminate unwanted polarity inversions. Fortunately there are tools, such as the Galaxy Audio Cricket, available for testing polarity of mics, mixers, effects devices, amps, cables, and speakers. These can save us hours (and dollars) when setting up and troubleshooting sound systems of any size. Here are few examples of what can go amiss.

**The Magical Pulpit!**

Major microphone manufacturers are known to make wireless mic's which are out of polarity. Also, historically, some mic manufacturers have not always followed the current PIN 2 (+) hot convention. Consider the following: A preacher delivers his sermon while moving about the altar. His voice is transmitted to the sound system via a wireless microphone. He approaches the pulpit and gestures grandly to drive his main point home. Unfortunately, the wired mic on the pulpit is out of polarity relative to his wireless mic. The impact of his message is "magically" lost in a sonic "tunnel!" A quick check of the system polarity would have revealed this situation before it became embarrassing. By reversing the polarity of either mic, this situation could have been avoided. When it comes to polarity, YOU NEED TO KNOW.

**That's some effect!**

Many mixers invert polarity in the effects, auxiliary, and monitor sends. If this condition is not corrected before the signal is returned to the mixer, a "hollow" sound can result. To compound the situation, many effects devices themselves invert polarity. Connect a polarity test set "transmitter" to a mixer input, using a cable which you have verified for correct polarity. Turn up the appropriate controls to feed the signal to the main, monitor, and effects buss outputs. Using another "verified" cable, connect the test set "receiver" to each of these outputs in turn and check for correct polarity. If the polarity of any of these is inverted, note the condition and check all of the subsequent devices in the signal chain. If all polarity inversions are not accounted for, the overall effectiveness of your system will be compromised. Again, when it comes to polarity, YOU NEED TO KNOW.

**Amplify this!**

Amplifier inputs are a notorious offender when it comes to polarity. The input interface wiring convention is PIN2 (+) hot (or TIP of 1/4"). Unfortunately, not all amp manufacturers follow this convention. If you have amps from more than one manufacturer, or sometimes, different models from a single manufacturer, you may encounter polarity inversion. On stage you might have cancellations occur between the monitor speakers and the front of house system.

**Speaking of speakers...**

Some major speaker manufacturers do not always follow standard polarity conventions when marking their drivers. One well known company marks their (+) terminal in BLACK rather than the audio standard RED. The following inverted polarity situations can arise between cabinets in a system or even within a multiple driver enclosure.

One "stack" is out of polarity with another.
Like drivers in the same enclosure are out of polarity.

Crossovers inherently cause phase shift, causing some strange problems at the crossover point. For instance, in a 2-way system, the woofer should test (+). Since 2-way systems often use a 12 dB/octave crossover, there will be 180 degrees of phase shift at the crossover point. To compensate for this, the tweeter will be wired OUT OF POLARITY with respect to the woofer. When using active crossovers, you may encounter 18 dB or 24 dB/octave crossover rates with 270 and 360 degrees of phase shift respectively. You NEED TO KNOW the crossover characteristics in order to determine proper polarity.

| Crossover Slope per Octave | Phase Shift |
|---|---|
| 6 dB | 90 degrees |
| 12 dB | 180 degrees |
| 18 dB | 270 degrees |
| 24 dB | 360 degrees |

**And what about SPLIT personality?**

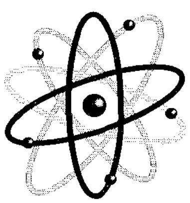
When your stage or recording studio setup calls for a transformer isolated splitter box to send signals from mics and instruments to the front of house, recording, and/or monitor consoles, you can assure correct polarity by using a Cricket. A continuity type of tester will not work in this situation.

In a large sound installation there can be literally hundreds of places where polarity can become inverted, much to the detriment of system performance. The Galaxy Audio Cricket can help you quickly identify and resolve these situations before they become "problems". For a limited time, the Cricket is available to Syn-Aud-Con graduates directly from Galaxy Audio at a special price. You may order by calling (800) 369-7768 and asking for the Syn-Aud-Con special.

*Submitted by Brock Jabara with Marc Chover of Galaxy Audio*

# Cycles, Circles and Triangles

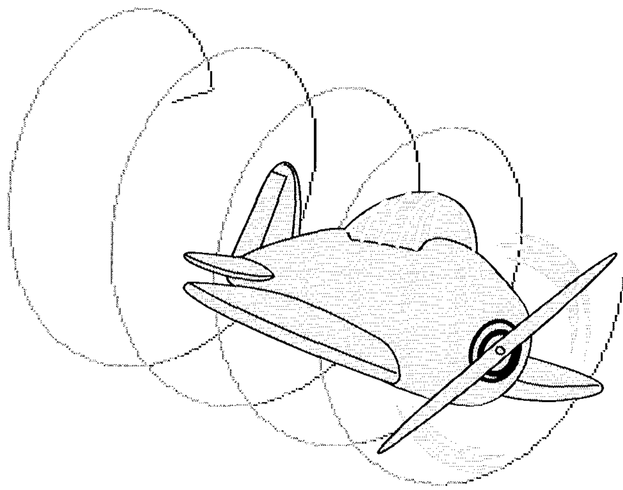
*Modeling the World Around Us*



Cyclic phenomena are fundamental to the world around us, and a closer look at cycles and circles will yield insight into the basic principles of audio and acoustics. This is of prime importance if one is to understand the workings of a sound system, or attempt to quantify the response of a system through measurement.

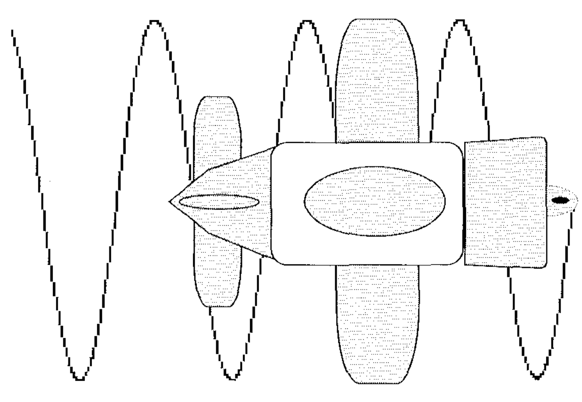
The circle is a unique shape, a line without beginning or end. It's circumference is related to it's diameter by a factor of 3.14, a relationship of such importance that it has it's own Greek letter, Pi. With the relationship between diameter and circumference established, it is possible to analyze cyclic phenomena and derive information about time-dependent functions, such as sound waves.

Great use of this diameter/circumference relationship is made in acoustic measurements. The circle is an ideal shape to describe cyclic phenomena, things that happen over and over again. This could be the rotation of the earth around the sun, an electron around a nucleus, or the hands of a clock. In audio and acoustics we use the circle to aid in describing the movement of a loudspeaker diaphragm, a microphone diaphragm, or the vibration of a molecule of air. To learn more about cycles and circles, lets use an example of rotational motion as a function of time and distance, the propeller on an airplane. Now, sound waves don't move through space like propellers, but the principles of cyclic motion and partitioning of energy between potential and kinetic are the same. The movement of the propeller is a function of the movement of the piston in the engine, so our model can also represent it's activity. An acoustic signal is also a function of the movement of a piston, the loudspeaker, and the resultant motion of the air particles pushed by it tell us a great deal about the loudspeaker that pushed them. The sound that we hear at a listener position in an auditorium is actually a "domino effect" that originates at the sound source (talker). The sound waves generated by the talker are converted into electromagnetic waves by the microphone. After being processed and amplified, they are converted back into acoustic waves by the loudspeaker. Even though the domain has alternated between acoustical and electrical, the same cyclic relationships exist between the components of the signal. Most loudspeaker response measurements are based on this "domino effect" information. Meanwhile, back to the propeller for a closer look at cyclic motion.

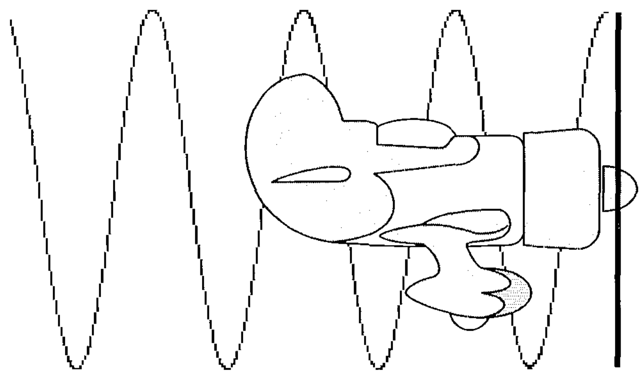


*Cyclic motion over time*

Let us mark one tip of the propeller, and track its motion over time. A three dimensional view reveals the propeller's motion as a corkscrew or spring. If this corkscrew were viewed only two dimensions at a time, the top view reveals the familiar sine wave and the side view the cosine wave. These two



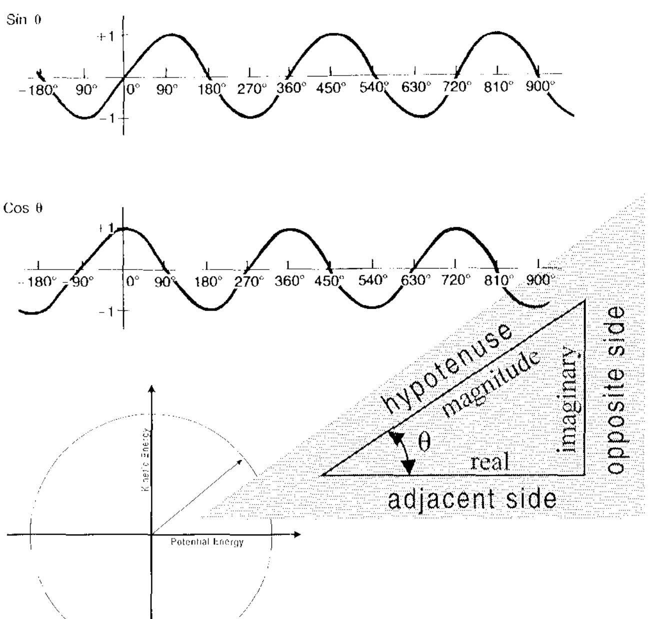
*Potential energy component*



*Kinetic energy component*

viewpoints are a look at the same event from different angles, in this case 90 degrees. Notice that the two waves look identical, except for their 90 degree offset. The top view represents the potential energy component of the piston, the energy due to its position. The side view represents the kinetic energy of the piston, its energy due to motion. It is interesting to note that if you only had one view, i.e., the potential or kinetic, the other could be found by shifting that information by 90 degrees. This is also true for the potential/kinetic energy relationship in loudspeakers, and the shifting process is known in instrumentation circles (no pun intended) as the Hilbert Transform. The measurement microphone records the pressure (potential energy) component of the acoustic signal. The kinetic component can be calculated by applying the Hilbert Transform to the potential component. Modern DSP chips and CPU's can do this very quickly.

If the propeller is viewed from the front, it now appears as a circle, and it becomes apparent that we are observing a cyclic process, and that the sine and cosine waves each describe one half of the total energy of the process.



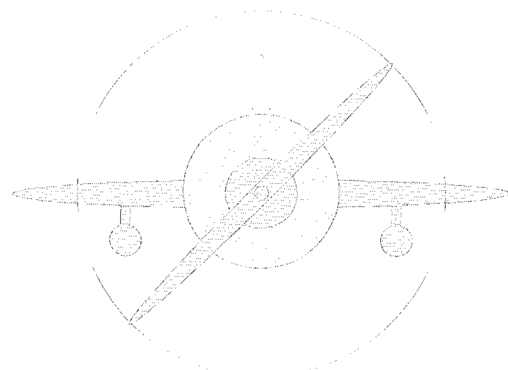

$$magnitude = \sqrt{(real)^2 + (imag)^2}$$

When a sine wave passes through zero, it simply means that there is no potential energy at that instant. A look at the cosine wave shows that all of the energy is kinetic. The circle shows that the total energy remains constant, but that it cycles continuously between potential and kinetic. It is the *combination* of these two types of energy that we experience in the world around us.

Electromagnetic waves do not require a medium to propagate through. Acoustic waves travel through mediums such as water, air, steel, etc. In sound, the movement of the loudspeaker causes movement of air molecules, which in turn causes movement of the eardrum. The movement of the eardrum is converted into the signals that our brain decodes and we perceive as sound.

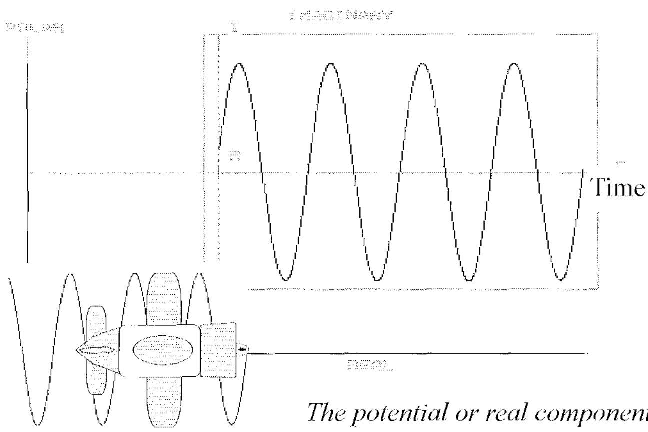
This "domino effect" is the basis of all acoustic measurements, allowing the potential and kinetic parts of the loudspeakers movement to be derived from the pressure variations at some distant point of observation.

Once the potential and kinetic parts of the energy under observation are known, it becomes possible to manipulate them mathematically into various displays, each having its own unique utility.



*A view of both potential and kinetic energy*

The tip of the propeller rotates at a fixed frequency. We can visualize the audio spectrum as 20,000 individual propellers, each rotating at a different frequency. The higher the frequency, the faster the rotation, or the more "cycles per second." If all of these propellers were placed on one shaft, and their rotation initiated at precisely the same instant in time, at some time later we could take a frontal "snap shot" of the propellers and observe their phase relationship to each other. At every one second interval all frequencies should be passing through zero phase. Sound waves propagate more like waves on water, and a measurement microphone is used to "sample" the wavefront as it passes the mic. If this were done for a perfect loudspeaker (which unfortunately cannot exist) using a broadband stimulus, the time relationship between the frequencies would be exactly the same as in the electrical signal that moved the loudspeaker. In effect, there would be a "dead heat" in the arrivals of all frequency components at the point of observation. This never happens in real life, and the resultant offset of each frequency (relative to the others) can occur for a variety of reasons and can be displayed in various ways. Heyser referred to the effect as time-delay distortion, the inevitable result of the signal propagating through a dispersive medium (the loudspeaker)[1]. The front view of the cyclic motion is called a Nyquist display, and for an acoustic signal each point on it represents the arrival of a specific discrete frequency. Some triangle math (see insets) allows the total magnitude and/or phase to be derived from the potential and kinetic (or real and imaginary, if you prefer) components. These components can be combined (with the help of Pythagoras) into the familiar frequency response plot, or the less familiar phase response plot. Each of these is useful for observing trends in the data, but it should be remembered that they are each derived from the same information. These interrelationships allow complex information to be derived from simple sound pressure measurements.

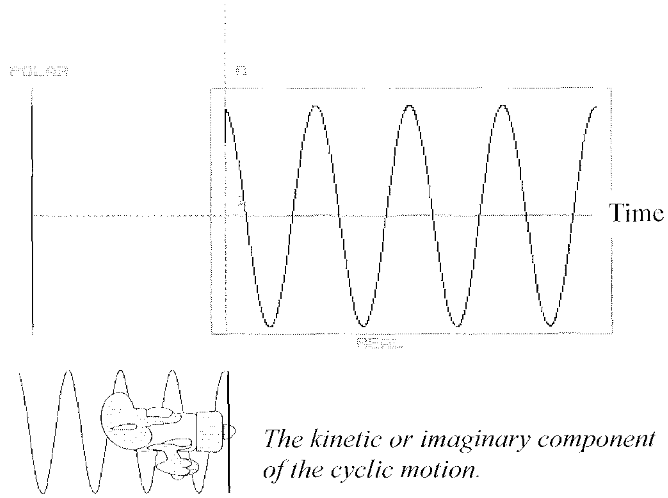



*The potential or real component of the cyclic motion.*

It is common practice in modern signal analysis to measure one energy component of the acoustic signal, and to derive the other with the Hilbert Transform. Once in hand, this information can be scrutinized from a variety of vantage points, including:

***magnitude response***
***phase response***
***impulse response***
***doublet response***
***Heyser Spiral***
***and others.***

The rotation of the propeller could be plotted as a function of time or as a function of frequency. In fact, if one of these were known, the other could be derived from it, since time and

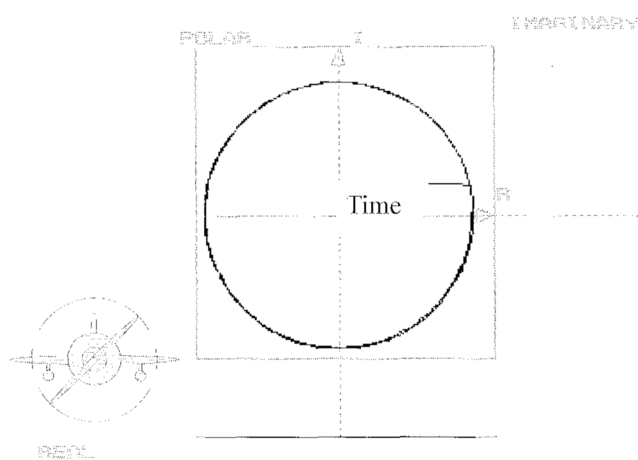



*The kinetic or imaginary component of the cyclic motion.*

frequency are the inverse of each other. If the propeller were rotating 1000 times per second, a frequency domain description would be 1000 cycles per second (Hz). In the time domain we could describe a single rotation as having a period of 1 millisecond. Time and frequency are different ways of describing the same event.

Just as the Hilbert Transform provided a map between potential and kinetic, the Fourier Transform provides a map between time and frequency. The impulse response (pressure vs. time) of a loudspeaker is the fundamental measurement of interest, but once acquired, it's information can be manipulated into a variety of displays.

Until this point we have been considering different views of continuous cyclic motion. Obviously, a moving propeller (or air particle) goes through many thousands of degrees of phase rotation as it propagates through the air. When making acoustic measurements, we are not interested in viewing this *absolute* phase rotation. We are interested in stopping the action at some point in time and space and observing the time (phase) relationship of the signals components as compared to their time (phase) relationship at the beginning of the process. Essentially, this involves comparing a snapshot of the signal at




*A view of potential and kinetic energy*

[1] Loudspeaker Phase Characteristics and Time-Delay Distortion: Part 1 JAES vol 17 no 1 January 1969, Richard C. Heyser

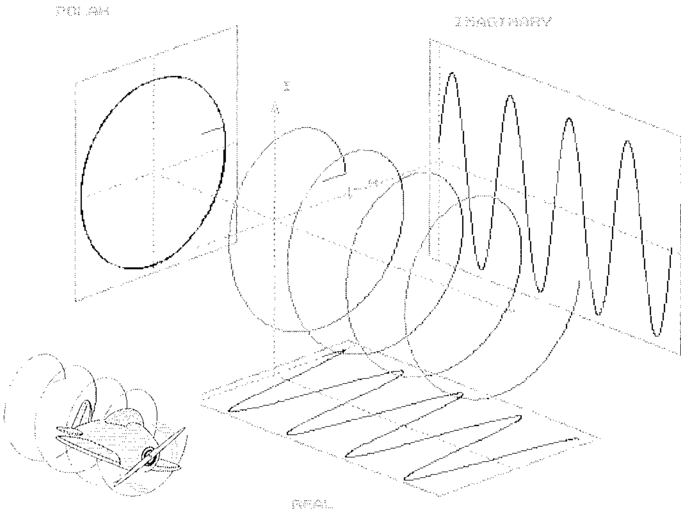
some point in time and at some distance to a snapshot of the original signal.

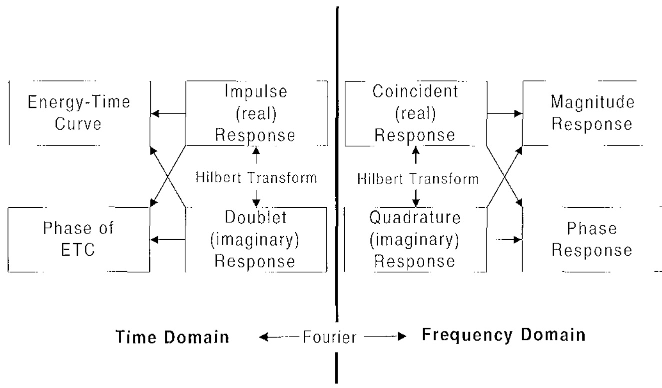
Observations of this type are commonplace in our lives. When you have your tires check for proper tread, the mechanic stops the phase rotation of the tire (turns off the engine!) and checks to see the toll taken by time and friction. A spinner on a child's game may go through many complete cycles of phase rotation when struck, but of concern is the relative phase position when it stops, not how many times it goes around. As we observe the clock during the workday, we seldom consider the amount of total time elapsed since the beginning of time (absolute time), but we are concerned with the position of the hands at the moment of observation and how it relates to our schedule (relative time).

This principle of "freezing the action" is vital to understanding acoustic measurements, and while the examples given may seem far removed from the subject, rest assured that in reality, making acoustic measurements is really no different than telling time or checking tires.

The Heyser Spiral is perhaps one of the most useful ways to view the data, at least from an academic point of view. In the time domain we see the pressure (or impulse) response, the doublet response (result of applying the Hilbert Transform to the impulse response), and the Nyquist response, which is actually a combination of the other two parts. The most common way of displaying this information is the Energy-Time curve, or ETC, which depicts the total energy density of the acoustic event. When this time data is ported over to the frequency domain via the Fourier Transform, it is observed as a coincident response and a quadrature response. These two responses are combined mathematically to provide the common frequency response and phase response plots.

We will be discussing all of this in more detail in future articles, but this should serve to stimulate the thought processes and cause one to ask "why", which is how any useful investigation begins. pb



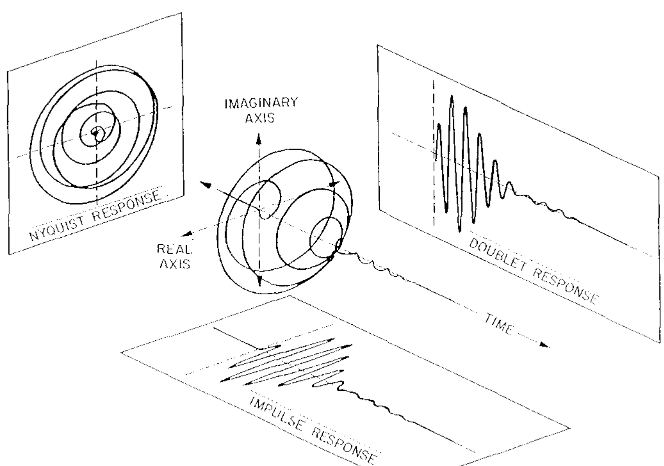




Common Analyzer Displays

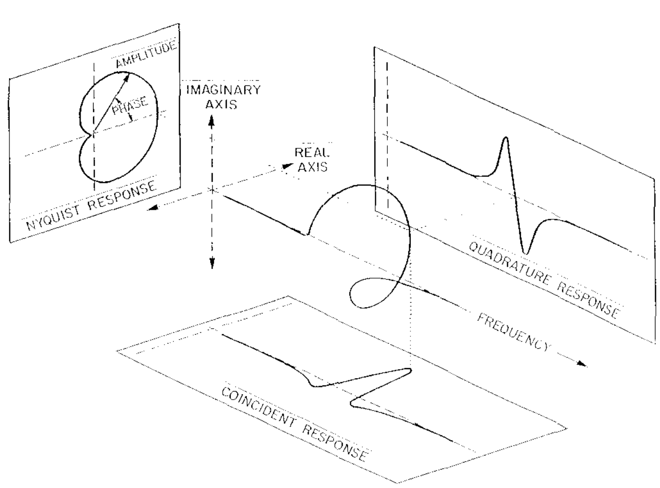
$$ETC = \sqrt{(real)^2 + (imag)^2}$$

$$Magnitude = \sqrt{(real)^2 + (imag)^2}$$

$$Phase = Tan^{-1}\frac{imag}{real}$$




The Time-Domain Heyser Spiral




The Frequency-Domain Heyser Spiral

# Lessons Learned from Pops and Cracks

A wise man once said that you should learn from your mistakes. An excellent opportunity arose when working on our test CD. The CD contains several tracks of pink noise bursts, designed to help estimate octave-band centered reverberation times of auditoriums (figure 1). Upon listening to the 125 Hz tracks, a peculiar "popping" sound was audible at the beginning and end of each burst. Using a wave editor to take a higher resolution look, it was apparent that there existed a very sharp slope at the beginning and end of each burst (figure 2). This was due to the method used to derive the signal. I thought that it might be interesting to take a look at the frequency content of the burst, and the tool of choice was the Spectrogram program reviewed in the newsletter earlier this year. Yes, the annoying "pops" are visible in the frequency domain, and Spectrogram clearly revealed the high frequency content of each.

This is an excellent example of why we like to look at data from several viewpoints before drawing conclusions from it. FFT analyzers allow both time and frequency domain views of the same data, and what is missed in one is often very apparent in the other. The failure of the noise track to begin at a zero crossing created a step function, which by definition will have a very broad band frequency content. Since the noise was band limited to the 125 Hz octave band, the high frequency information of the step function was very audible.

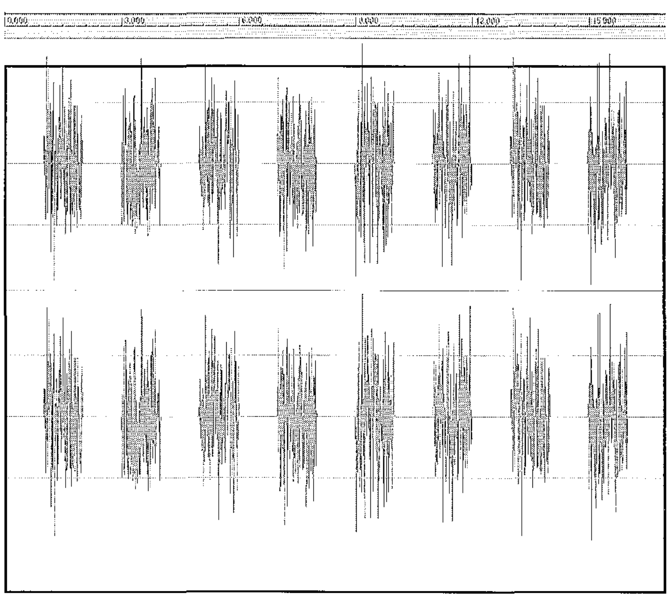


*Figure 1. Octave-band noise bursts over time*

The days of the phonograph record (what's that?) have passed, but most of us remember the pops and cracks that our LP's acquired with age. Scratches and dirt in the record grooves certainly imitated "step functions" and manifested their high frequency content upon each playing.

The solution for our test CD was to modify the slope with the wave editor, a luxury that didn't exist for our phonograph records (who said digital isn't better?).

A special thank you to Richard Clark of Autosound 2000, who pointed out the problem and the solution. pb

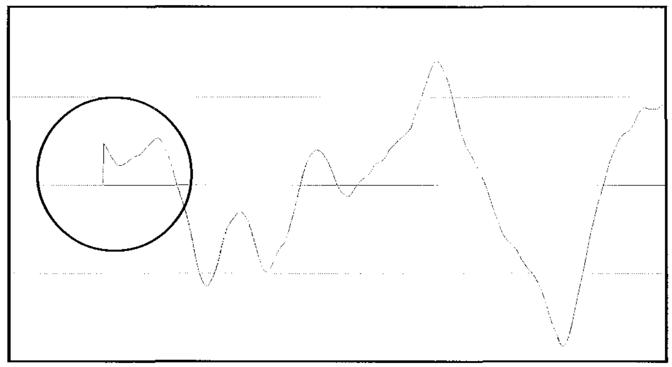


*Figure 2.*

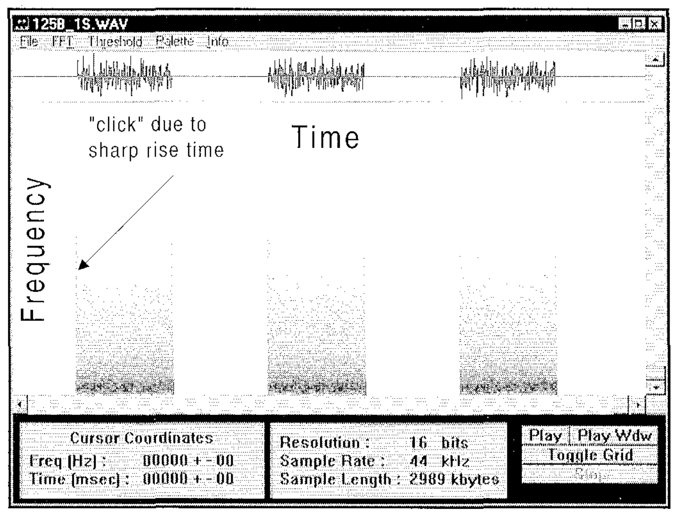



*Figure 3. showing high-frequency content of step.*

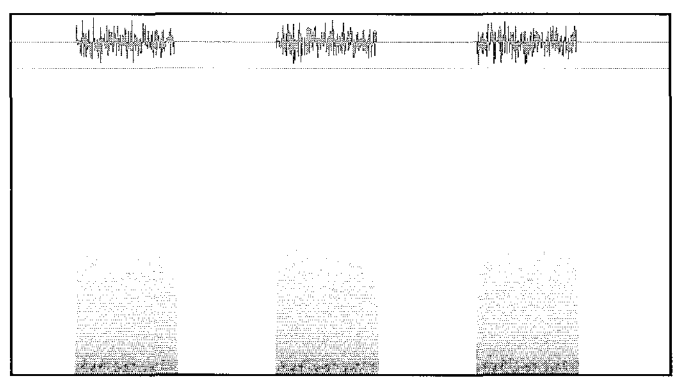


*Figure 4. Response after correction.*

# Limitations on Line Length Due to Cable Capacitance

When two electrical conductors are in close proximity to each other, there is capacitive coupling between them. This can be thought of as a frequency-dependent short between the conductors. As frequency gets higher, more of the signal is shorted to the other conductor (usually ground) and never makes it to the input of the next device. This pesky parameter pops up in many electronic applications, from PC board design to Local Area Networks, but in the audio world it becomes particularly bothersome in long cable runs. Capacitance that is inherent to shielded cable varies widely, and can have a profound effect on signal fidelity.

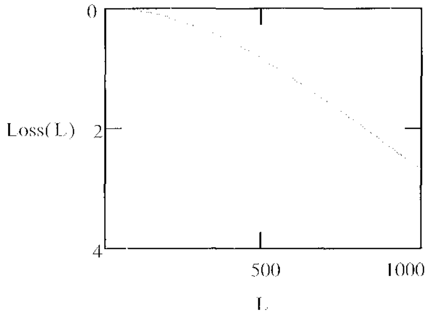
How much cable can we put between the mic and mixer input without significant high-frequency loss? This is a question to ask at the drawing board stage of a project, and this issue's Mathcad Quarterly will help answer that question. From the equation below, it is apparent that several ingredients go into the solution of this problem. It helps to have very low output impedances, and very low-capacitance cable. Try some different values in the equation and see what you get.

To run this template, you will need Mathcad from Mathsoft. The template can be downloaded at no charge from our Internet Website. pb

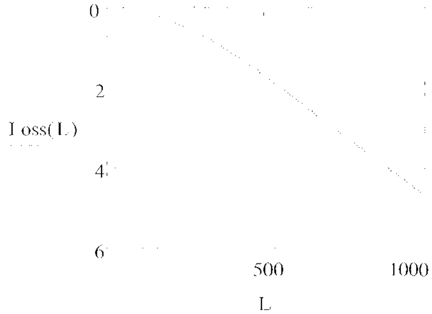
## Calculating Cable Capacitance Effects

| | | |
|---|---|---|
| $R_S := 200$ | Ohms | Source resistance of driving device |
| $R_L := 2500$ | Ohms | Input resistance of driven device |
| $f := 20000$ | Hz | Frequency of interest |
| $C := 40 \cdot 10^{-12}$ | farads | Capacitance per foot of cable |
| $L := 1 \ldots 1000$ ft | | Length of cable |

$$\mathrm{Loss}(L) := 20 \cdot \log\left(\frac{R_L + R_S}{\sqrt{\left(R_S + R_L\right)^2 + \left(R_S \cdot R_L \cdot 2 \cdot \pi \cdot f \cdot C \cdot L\right)^2}}\right)$$




*Loss at 20 kHz as a function of line length for 40 pF/ft cable.*




*Loss at 20 kHz as a function of line length for 60 pF/ft cable.*

# A Lesson on Waves from...

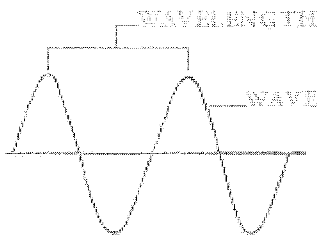
## THE WAY THINGS WORK
DAVID MACAULAY
HMCo



### GLOSSARY

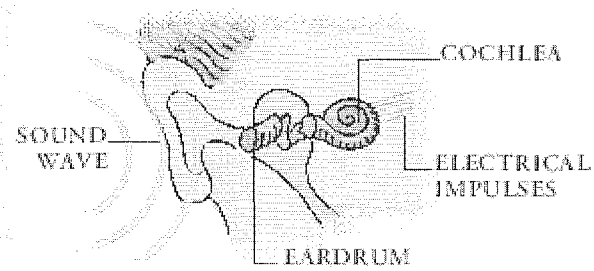
#### WAVELENGTH

Wavelength is the distance between two adjacent wave crests in a regular wave pattern. For example, the wavelength of waves in water would be the distance from crest to crest.




Wave

### HEARING SOUND

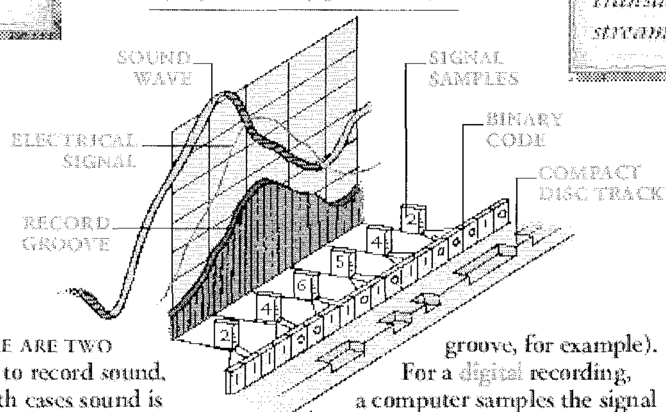



Sounds change the pressure in our ears, making the eardrum vibrate. Small bones behind the eardrum amplify the vibrations and transmit them to the fluid-filled cochlea. There the vibrations move tiny hairlike cells, sending electrical impulses to the brain.

**SOUND WAVE**
*The curve shows the changes in pressure and energy in a sound. We hear these changes as different notes and volumes.*

**SIGNAL SAMPLES**
*To make a digital recording the signal is sampled by a computer, translating it into a stream of numbers.*

### RECORDING SOUND




**ELECTRICAL SIGNAL**
*A microphone is used to convert sound waves into a continuously varying electrical signal.*

**COMPACT DISC TRACK**
*For a compact disc recording, the signal is converted into binary code. This is recorded as a series of pits on the disc's surface.*

THERE ARE TWO WAYS to record sound. In both cases sound is first converted into an electrical signal. An analog recording copies the signal's shape (as varying magnetism on tape, or as a record groove, for example). For a digital recording, a computer samples the signal thousands of times a second. The numbers are translated into binary code and stored, for example, on a compact disc.

### GLOSSARY

#### ANALOG

An analog electrical signal or recording varies continuously in the same pattern as the change which it represents. The ripples in the groove on a vinyl record, for example, are a physical copy of a sound wave.

Digital

### GLOSSARY

#### DIGITAL

A digital signal is one in which a changing pattern, such as a sound wave, is represented as a series of numbers or digits. The sound signal on a compact disc, for example, is recorded as a sequence of binary numbers.

Analog

# Compensating for Air Absorption When Using the Sabine Equation

The well-known Sabine equation is a useful tool for estimating the reverberation time of enclosed spaces. Sabine arrived at the formula empirically, by many hours of testing in many spaces. The formula is as follows:

$$RT_{60} = \frac{0.161V}{S\bar{a}} \qquad RT_{60} = \frac{.049V}{S\bar{a}}$$

Metric English

where V is the volume of the space
S is the interior surface area
a is the average absorption coefficient

The term for absorption in the denominator represents the total absorption present in the space. The total absorption can be further subdivided into the absorption of the surface materials and the absorption from the air itself, which can become a factor in large spaces. With that in mind, let us describe the total absorption with separate terms for the surface and air absorption. According to Baranek:

$$RT_{60} = \frac{0.161V}{S\bar{a} + 4mV}$$

where m is the atmospheric attenuation coefficient.

This multiplier, which is both temperature and humidity dependent, can be acquired from the chart, which can be found in the Audio System Designer. For example, the contribution of air to the total absorption at 68°F and 50% humidity can be found as follows:

$$Value\ from\ chart = 1.0$$

$$Conversion\ to\ m\ value = \frac{1}{434}$$

$$m\ value = 0.0023$$

$$for\ a\ room\ of\ 7000\,m^3\ (250{,}000\ ft^3):$$

$$S\bar{a}\ from\ air = 4mV = 4(.0023)(7000)$$

$$= 64\ Sabins\ (metric)\ or\ 693\ Sabins\ (English)$$

*The Audio System Designer from Peter Mapp and Mark IV is a valuable resource for system design data.*

In the previous example, the contribution of absorption from air is about 693 English Sabins. If the measured 2000 Hz $RT_{60}$ of the space is 2.5 seconds, then the total absorption could be calculated as follows:

$$S\bar{a} = \frac{0.049V}{RT_{60}} = \frac{0.049(2.5)(10^3)}{2.5}$$

$$S\bar{a} = 3062\ \text{Sabins}$$

This represents the total absorption present, including air absorption. To find the absorption contribution of the room surfaces, we must subtract the air absorption found previously from this value.

3062 - 693 = 2369 Sabins from the room surfaces

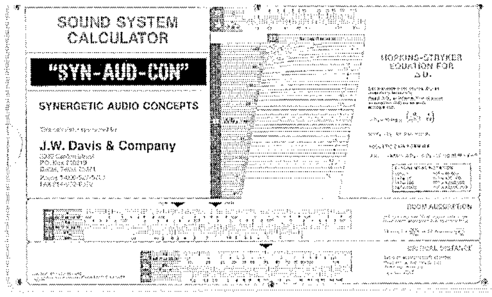
In this example, the 2 kHz absorption contribution of the air is the equivalent of about 170 people at 4 Sabins each. This is important to remember when deriving the surface material absorption present from $RT_{60}$ measurements.

**Atmospheric Attenuation Coefficient in dB/100 m for an Atmospheric Pressure of 1.013 x $10^5$ Pa (One Standard Sea-Level Atmosphere)**

| *Temperature* | *Relative humidity, per cent* | *Frequency* 125 | *250* | *500* | *1000* | *2000* | *4000* |
|---|---|---|---|---|---|---|---|
| | 10 | 0.09 | 0.19 | 0.35 | 0.82 | 2.6 | 8.8 |
| | 20 | 0.06 | 0.18 | 0.37 | 0.64 | 1.4 | 4.4 |
| 30°C | 30 | 0.04 | 0.15 | 0.38 | 0.68 | 1.2 | 3.2 |
| (86°F) | 50 | 0.03 | 0.10 | 0.33 | 0.75 | 1.3 | 2.5 |
| | 70 | 0.02 | 0.08 | 0.27 | 0.74 | 1.4 | 2.5 |
| | 90 | 0.02 | 0.06 | 0.24 | 0.70 | 1.5 | 2.6 |
| | 10 | 0.08 | 0.15 | 0.38 | 1.21 | 4.0 | 10.9 |
| | 20 | 0.07 | 0.15 | 0.27 | 0.62 | 1.9 | 6.7 |
| 20°C | 30 | 0.05 | 0.14 | 0.27 | 0.51 | 1.3 | 4.4 |
| (68°F) | 50 | 0.04 | 0.12 | 0.28 | 0.50 | 1.0 | 2.8 |
| | 70 | 0.03 | 0.10 | 0.27 | 0.54 | 0.96 | 2.3 |
| | 90 | 0.02 | 0.08 | 0.26 | 0.56 | 0.99 | 2.1 |
| | 10 | 0.07 | 0.19 | 0.61 | 1.9 | 4.5 | 7.0 |
| | 20 | 0.06 | 0.11 | 0.29 | 0.94 | 3.2 | 9.0 |
| 10°C | 30 | 0.05 | 0.11 | 0.22 | 0.61 | 2.1 | 7.0 |
| (50°F) | 50 | 0.04 | 0.11 | 0.20 | 0.41 | 1.2 | 4.2 |
| | 70 | 0.04 | 0.10 | 0.20 | 0.38 | 0.92 | 3.0 |
| | 90 | 0.03 | 0.10 | 0.21 | 0.38 | 0.81 | 2.5 |
| | 10 | 0.10 | 0.30 | 0.89 | 1.8 | 2.3 | 2.6 |
| | 20 | 0.05 | 0.15 | 0.50 | 1.6 | 3.7 | 5.7 |
| 0° | 30 | 0.04 | 0.10 | 0.31 | 1.08 | 3.3 | 7.4 |
| (32°F) | 50 | 0.04 | 0.08 | 0.19 | 0.60 | 2.1 | 6.7 |
| | 70 | 0.04 | 0.08 | 0.16 | 0.42 | 1.4 | 5.1 |
| | 90 | 0.03 | 0.08 | 0.15 | 0.36 | 1.1 | 4.1 |

The Value of m in the 4mV reverberation time formulae is obtained by dividing the above Values by 434

# Estimating %Alcons



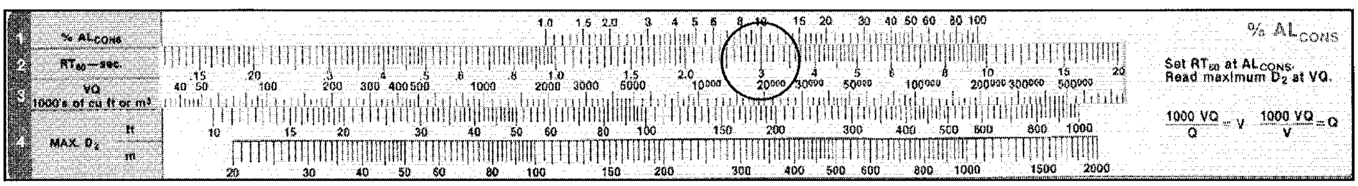

The %Alcons equation from V.M.A. Peutz is still our best "drawing board" tool for predicting speech intelligibility. Mr. Peutz was able to quantify the factors involving intelligibility in reverberant rooms through years of measurement and correlation in real spaces. The Syn-Aud-Con Slide Rule provides a quick and easy way to execute the equation given some basic room parameters, which include:

V - the room volume in cubic feet or meters
$D_2$ - the distance to the seat under consideration
Q - the directivity factor of the source
$RT_{60}$ - the reverberation time of the space

With these parameters in hand, you may proceed as follows:

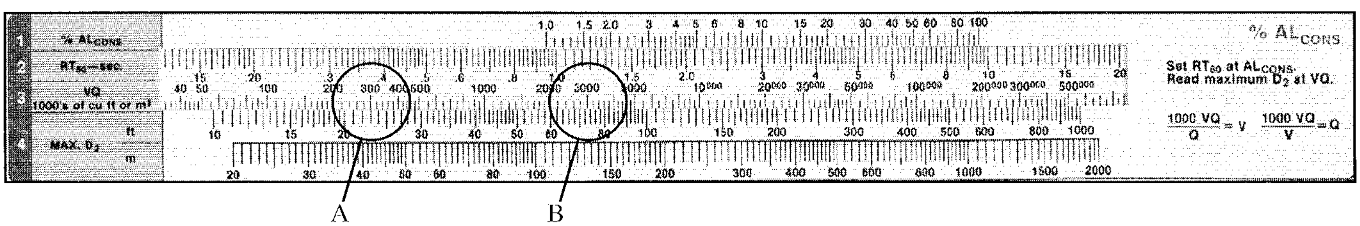
Given a room with an internal volume of 300,000 ft$^3$, an $RT_{60}$ of 3 seconds, and a target %Alcons of 10%, establish the relationship between loudspeaker directivity (Q) and maximum listening distance.

1. Referring to the %Alcons section of the slide rule, align 3 seconds on scale 2 ($RT_{60}$), with 10% on scale 1.



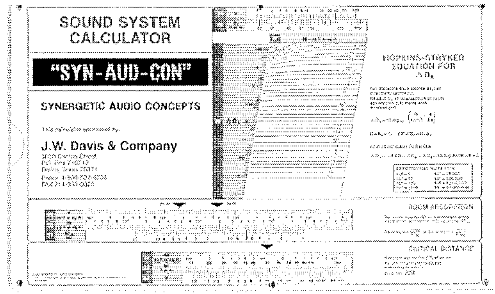

2. Note that scale 3 is the product of the room volume and directivity factor, and that the volume is in thousands of cubic feet. This may seem confusing, but it actually makes things easier if you proceed as follows:

Since the given room volume was 300,000 ft$^3$, the simplest thing to do is to initially consider Q to be unity (omnidirectional source). Since the volume is in thousands of cubic feet, the large number 300,000 will be reduced to 300 and located on scale 3. With Q = 1, the maximum distance for 10% Alcons is 23 feet (circle A). Not surprising with a low-Q source.




If Q were increased to 10 (a typical 90 x 40 degree horn), the maximum listening distance becomes 72 feet. In fact, the maximum listening distance for any directivity factor can be estimated by multiplying 300 by Q. Real-world Q values will range between 1 and 50. If multiple loudspeakers are required to cover different areas in the same room, this can be accounted for by scaling the $RT_{60}$ by the total number of loudspeakers required for coverage at 2 kHz. For example, if two loudspeakers were required (same type, equal power, not covering the same area), this would mathematically be the same as halving the room absorption, which would in turn double the $RT_{60}$. We could then set the $RT_{60}$ to six seconds and repeat the above process, seeing the effect on intelligibility of the increased number of devices. The maximum listening distance for a directivity factor of 10 is reduced from 72 feet to about 32 feet. This is why we emphasize minimizing the number of acoustic sources in reverberant spaces.

# Finding Acoustic Wavelengths




One of the most important concepts in sound system design is the fact that soundwaves in air have a physical size. In our training seminars, we teach and stress the use of acoustic wavelengths in sound system design. The concept is so fundamental that virtually no aspect of the acoustical performance of the sound system escapes its influence. Just a few of the things that are wavelength dependent are:

**Room Modes**
**Loudspeaker Directivity**
**Absorption**
**Diffusion**
**Reflection**

The list could go on and on. With this in mind, we will used the Syn-Aud-Con slide rule to find an unknown wavelength from one that we know.

The first step is to establish a reference wavelength. The math is as follows:

$$T = \frac{1}{f}$$

*where:*

$T = time$

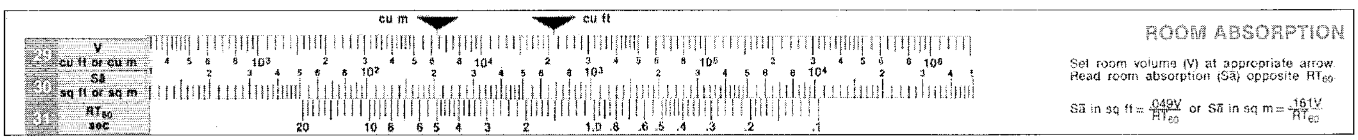
$f = frequency$

$$\lambda = T \times 1130$$

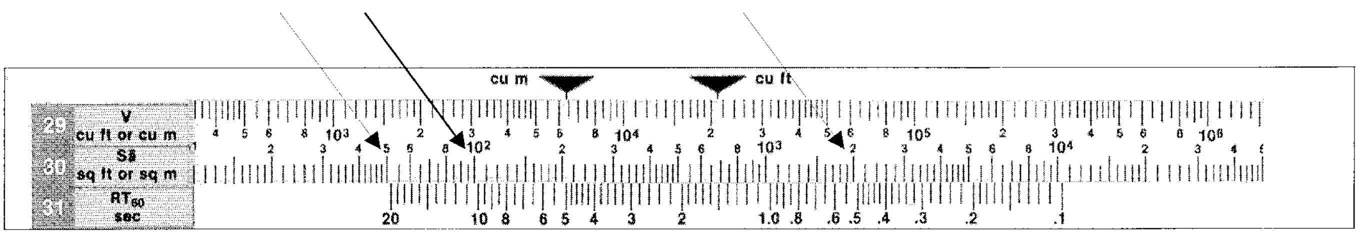
*where:*

$\lambda$ *is wavelength*

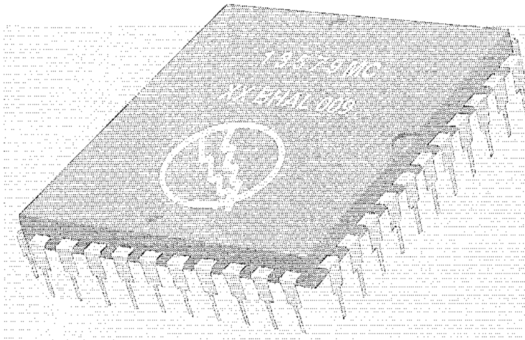
Using these relationships, the wavelength of 1000 Hz becomes 1130/.001 or 1.13 feet. We will use the Room Absorption section of the slide rule to find other wavelengths from this relationship.



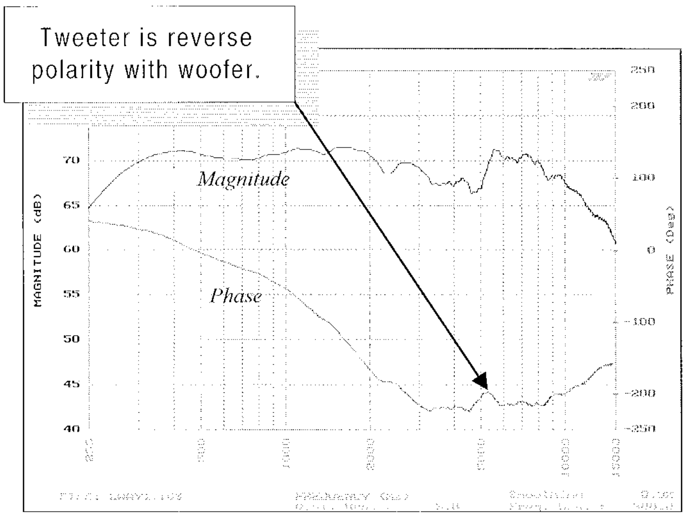

1. Referring to the Room Absorption section of the slide rule, let scale 30 represent frequency and scale 31 represent feet.
2. Align 1000 Hz ($10^3$) from scale 30 with 1.13 ft on scale 31.
3. Read any other wavelength at any other frequency.



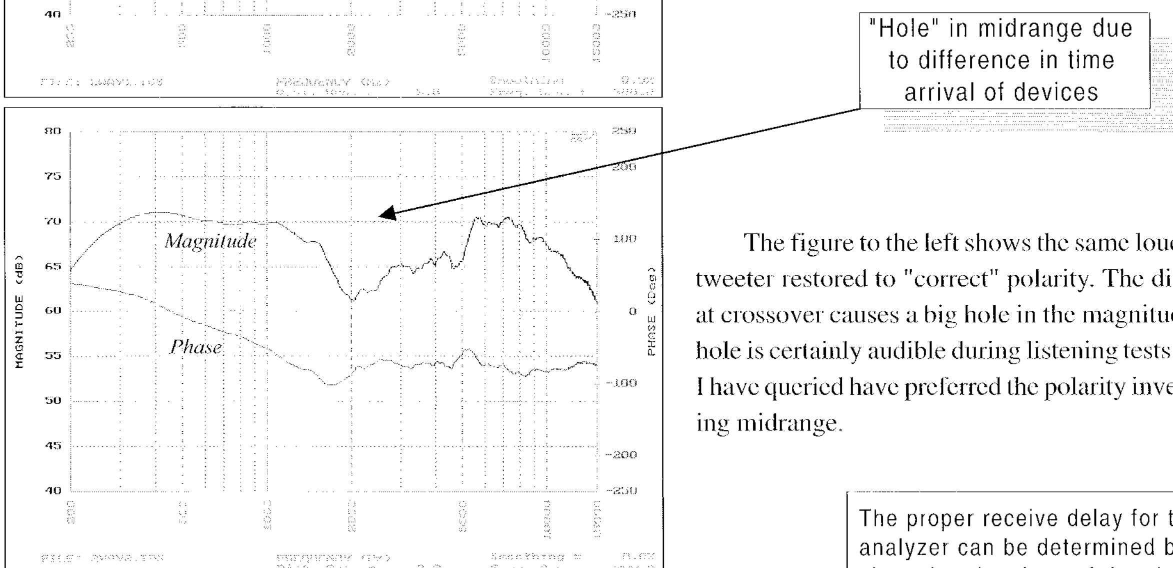
From this example, we can see that the wavelength of 100 Hz is 11.3 feet, 50 Hz is 20 feet, 2000 Hz is 0.5 feet and so on. Sound becomes larger as frequency gets lower. This explains why high frequencies are very easy to absorb and reflect, while low frequencies tend to be omnidirectional and difficult to contain.



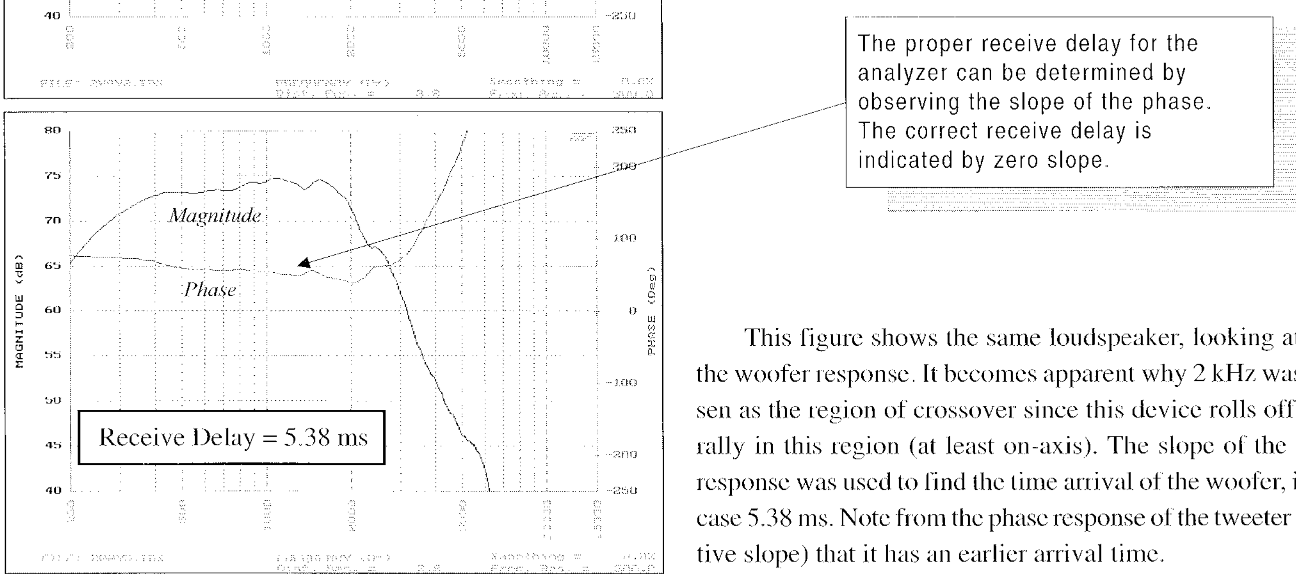
# Making a Good System Better with DSP




A high-quality two-way loudspeaker system for A/V and foreground music applications consists of a dome tweeter and 4.5" woofer. The unit comes configured with a passive crossover. The performance at the 2 kHz crossover region has a very respectable magnitude response, but the polarity of the dome tweeter had to be reversed to achieve it. Whether or not this polarity inversion is audible has been the subject of many ongoing debates. Audible or not, it is certainly measurable!

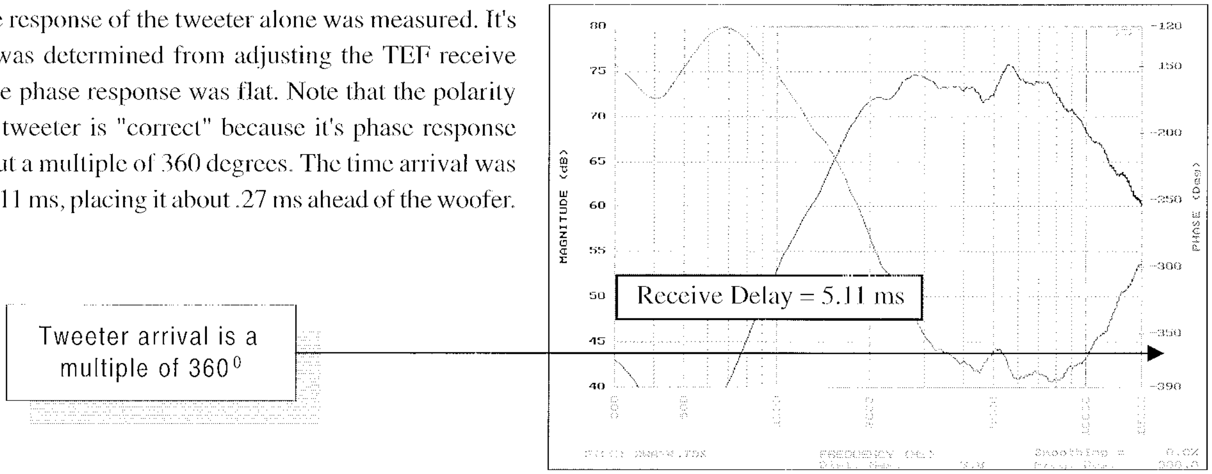



The figure to the left shows the same loudspeaker with the tweeter restored to "correct" polarity. The difference in phase at crossover causes a big hole in the magnitude response. This hole is certainly audible during listening tests. All listeners that I have queried have preferred the polarity inversion to the missing midrange.

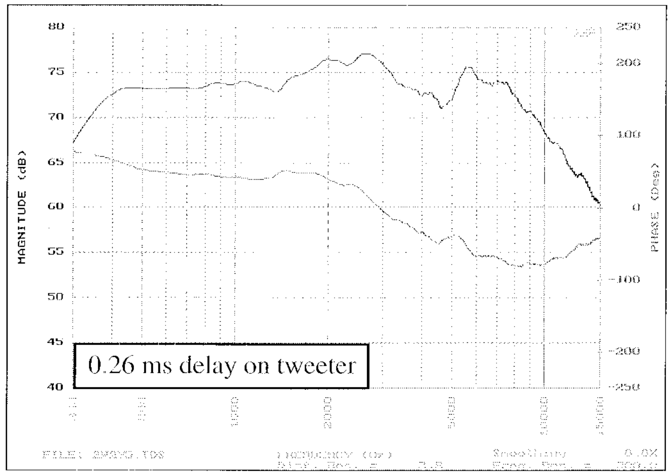


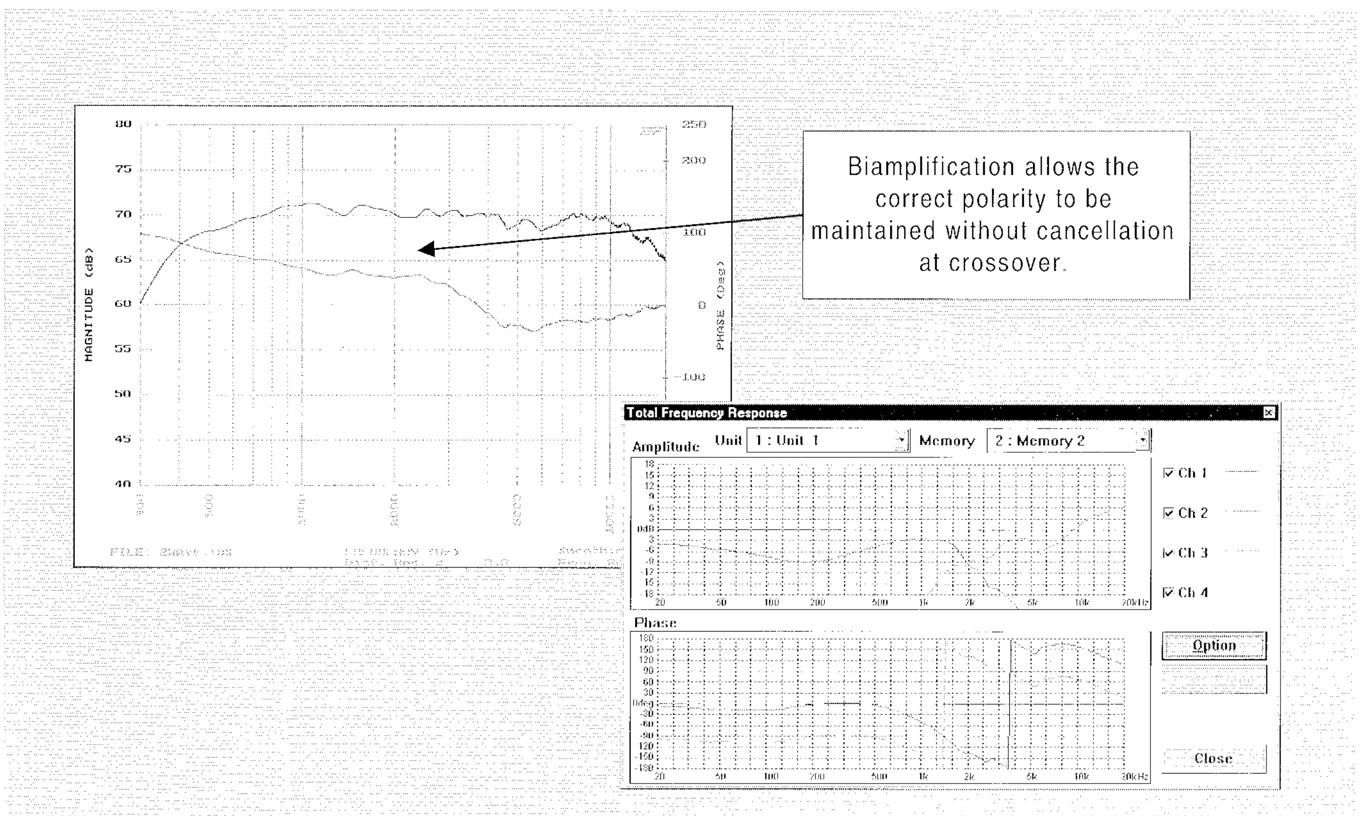

This figure shows the same loudspeaker, looking at only the woofer response. It becomes apparent why 2 kHz was chosen as the region of crossover since this device rolls off naturally in this region (at least on-axis). The slope of the phase response was used to find the time arrival of the woofer, in this case 5.38 ms. Note from the phase response of the tweeter (positive slope) that it has an earlier arrival time.

Next, the response of the tweeter alone was measured. It's arrival time was determined from adjusting the TEF receive delay until the phase response was flat. Note that the polarity of the of the tweeter is "correct" because it's phase response averages about a multiple of 360 degrees. The time arrival was found to be 5.11 ms, placing it about .27 ms ahead of the woofer.

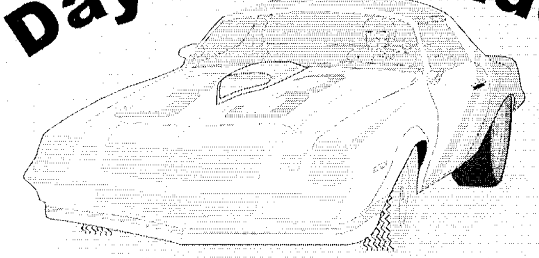



The TOA DacSYS processor was used to provide a 2 kHz crossover point, as well as to provide the needed delay to the tweeter. The nearest delay to optimum was .26 ms. This setting allows the woofer and tweeter to operate with the same polarity, without the notch at crossover. At this point we turn to the parametric equalizer function of the DacSYS to smooth the magnitude response. The graphs below show the final loudspeaker response, as well as the transfer function of the DacSYS processor. The moral of the story? If price is no object, improvements (at least measurable ones, if not audible ones) can be made in about anything. pb



# Days of Thunder

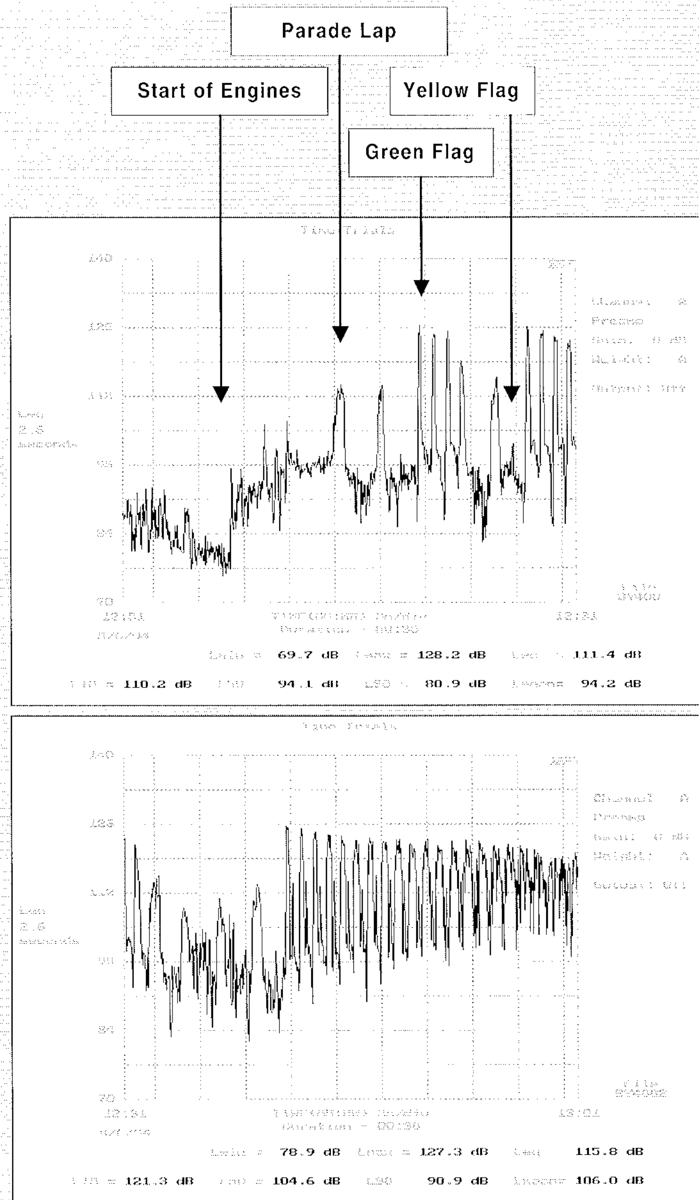


The Indianapolis Motor Speedway is always an interesting place to do measurements. This past August I measured the noise levels present during the Brickyard 400 Stock Car Race. As you can see, this isn't a great place to "read a book." The test microphone was about 100 feet from the track. The TEF NLA software allowed the entire race to be measured in 30 minute "slices." The graph to the right includes the opening ceremonies, aircraft fly-over, parade laps, and initial race laps. At best, the sound system is usable between cars and during yellow flags. While it would certainly be possible to exceed the level of the cars, I doubt that it would win John Royer many friends. Use your "Syn-Aud-Con Slide Rule" to find the maximum level in the pits (about 30 feet from the track). Mathematically, it would be approximately:

$$L_p = 128 + 20\log\frac{100}{30} = 138.5dB$$

Hearing protectors are commonplace at the track, and are included with every race program purchased. I found it useful to plug one ear, and then turn that one to the track as the cars passed. Care must be taken in the garage area, since the unwary passerby can be just a few feet from the rear window exhaust pipe of a stock car under test.

During the race we used the Communications Company SP-1 Speech Intelligibility Processor to enhance the speech. It worked quite well when the cars were not passing by.

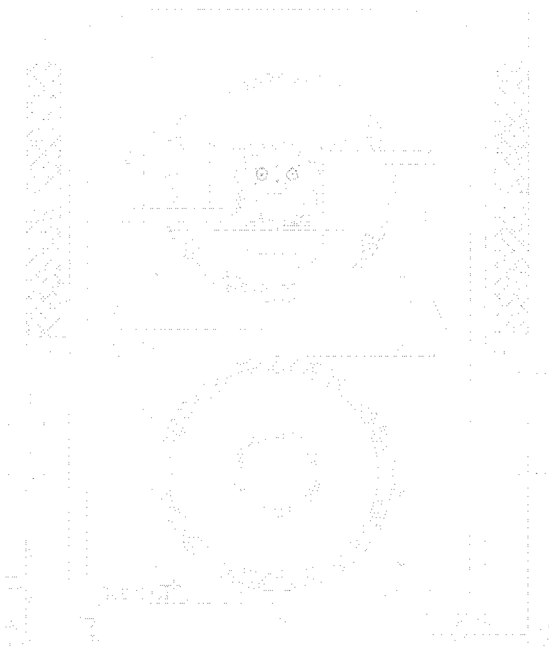



*Much can be determined about the race from the sound record. As the cars disperse over time, the level drops. This is the same principle that describes the operation of acoustic diffusors.*

"The Syn-Aud-Con Newsletter is the "water cooler chat" of a very distributed industry."

*Barry McKinnon*

# Community Data Available on CDROM or in Binder



It takes a lot of data to describe a loudspeaker, and even more to describe an entire product line. A typical consultant or contractors office wall looks like a shrine to the "binder god." I have long wished that all of this data could be available on CD so that I can have my wall back for pictures, windows, dart boards and life's other essentials.

Community has blazed a trail by providing their data on CDROM, in addition to the standard (really nice) binder. The CD has product drawings in several formats (including Autocad), as well as specification sheets in Word 6.0 and .PDF (Adobe Acrobat) format, complete with the Acrobat 2.1 reader. Current product drawings and spec sheets are no further away than the computer. Your customers will be impressed with laser quality data sheets, rather than the copies that sometimes had to be used on that "last minute" project.

We hope that other manufacturers follow Community's lead and archive their data on CD. But then what would we do with all of that wall space? *pb*

EAW

EASTERN ACOUSTIC WORKS

**Visit the site at:**
**http://www.eaw.com**

# Information Abounds on EAW's Website

If the information superhighway can have a traffic jam, EAW may be a contributing factor. Their home page on the World Wide Web is a graphic experience, and takes HTML publishing technology to it's limits. EAW's dealers will find the site quite useful for downloading product information, including drawings and specification sheets. Others may want to visit just to see what can be done on the Web. In either case, be sure that you have a fast modem and some extra time to enjoy the show.

# Professional Services

Acoustical Consultants are provided a listing in this section. 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.

KMK
7 Holland Avenue
White Plains, New York 10603
Telephone (914) 761-8595

**David L. Klepper**
7 Holland Ave.
White Plains, NY 10603
914-761-8595

WILLIAM THOMAS STREIBLE

2041 N. Commonwealth Ave.
Suite 107
Hollywood, CA 90027

**EUGENE T. PATRONIS, JR.**
PH.D.
School of Physics
Georgia Tech
Atlanta, GA 30332
404-8945237

ACOUSTICAL DESIGN GROUP INC

**Ian R. Wolfe**
6700 Squibb Rd. Suite 204
Mission, KS 66202
913-384-1261
913-384-1274 Fax

ACOUSTICAL ENGINEERS INC.
Consultants

**Richard J. Fullmer, P.E.**
1864 S. State #270
Salt Lake City, UT 84115
801-467-4206

**Jordan Audio Consultants**

**Gary Jordan**
RR1 Box 625
Joplin, Missouri 64801
417-623-7286 Email JSBP@aol.com

LEE SOUND DESIGN
Consulting Engineering - Construction Management

**D. Wayne Lee, P.E.**
1474 Stephens Drive, NE
Atlanta, GA 30329-3745
404-633-8590

LEWITZ AND ASSOCIATES
AUDIO AND ACOUSTICAL CONSULTANTS

**Joel A. Lewitz, P.E.**
1505 Bridgeway Suite 206
Sausalito, CA 94965
415-332-3434
415-332-6340 Fax

*Sound Visions Consulting*

**Ray A. Rayburn**
1184 W. Corporate Dr.
Arlington, TX 76006
817-640-7300
817-633-5920 Fax

MICHAEL R. YANTIS ASSOCIATES, INC., P.S.
consulting in acoustics and vibration

**Todd S. Welti**
12509 Bel-Red Road, Suite 203
Bellevue, WA 98005
206-454-4283
206-454-0759 Fax

PAOLETTI Associates

**Kurt M. Graffy**
40 Gold St.
San Francisco, CA 94133
415-391-7610
415-391-0171 Fax

WILLIAM W. SETON & ASSOCIATES
ELECTRO-ACOUSTIC CONSULTING SERVICES
Specification • Evaluation • Testing • Training

**William W. Seton**
2006 Naudain Street
Philadelphia, PA 19146
215-732-9001

Towne, Richards & Chaudiere, Inc.
Consultants in Sound and Vibration

**Herbert T. Chaudiere**
105 NE 56th St.
Seattle, WA 98105
206-523-3350

The Greenbusch Group, Inc.
Acoustical and Mechanical Engineering Consultants

**Julie A. Wiebusch**
563 NE Ravenna Blvd.
Seattle, WA 98115
206-524-0593
206-524-0630 Fax

LARRY ELLIOTT ASSOCIATES·LIMITED

**Larry J. Elliott**
22A Tramway Rd. Beachhaven
PO Box 66-050 Auckland 10 New Zealand
64-9-482-0772
64-9-483-4551 Fax

AMERICAN ACOUSTICAL ASSOCIATES

**Kenneth B. Scott. P.E.**
P.O. Box 987
Powell, OH 43065
614-889-2899

**Jim Faber**
13714 Gamma Rd. Suite 110
Dallas, TX 75244
214-934-3700
214-934-3720 Fax

WJHW
Wrightson, Johnson, Haddon & Williams, Inc

Orchestral Arts Inc
ORCHESTRAL ARTS
Acoustic and Performing Technologies

**Dale Fawcett**
227 Cosburn Ave. No. 706
Toronto, Ontario M4J 2L6 Canada
416-469-4478

The Audio Systems Group Inc.

**Jim Brown**
4875 N. Ravenswood
Chicago, IL 60640
312-728-0565

OLSON SOUND DESIGN

**Bruce C. Olson**
8717 Humbolt Ave. North
Brooklyn Park, MN 55444-1320
612-493-5835
612-493-4491 Fax

PMK Pelton Marsh Kinsella

**David E. Marsh**
7950 Elmbrook Dr. Suite 100
Dallas, TX 75247-4951
800-229-7444
214-951-7408 Fax

**You might be a sound guy if...**

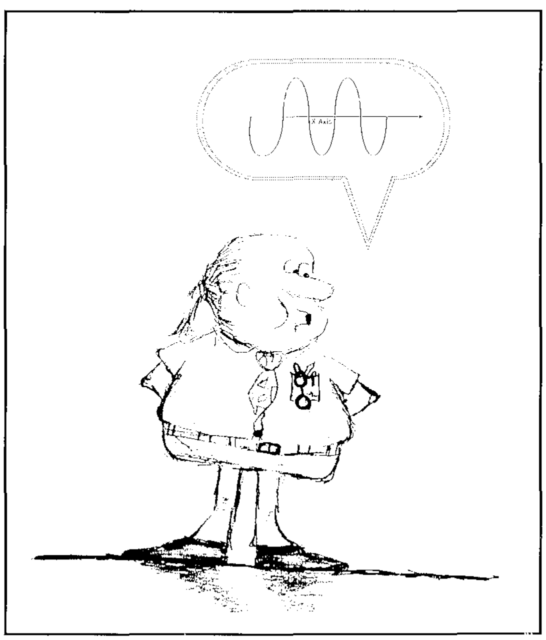
Your favorite song is "pink noise."

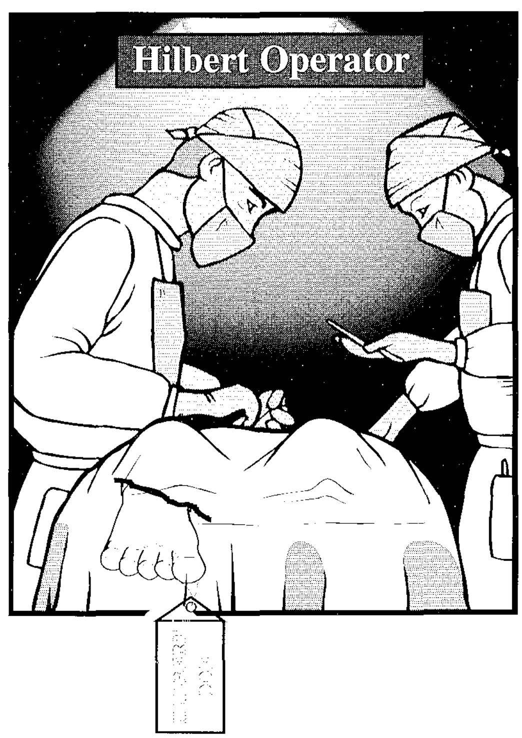
You find yourself checking loudspeaker density when eating in restaurants.

Your hi-fi system is made up of leftover equipment from your band's PA system.

You hang-out behind the mix position at concerts.

**Dr. Audio** **by Ashley Brown**





# SYN-AUD-CON SPONSORS...

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

| | | |
|---|---|---|
| JBL UREI ELECTRONIC PRODUCTS | **JBL, Inc.**<br>8500 Balboa Blvd.<br>Northridge, CA 91329<br>818-894-8850 Phone<br>818-830-1220 Fax | Manufacturer of horns, loudspeakers, drivers, enclosures, electronics, CADP2 system design program for Windows. |
| JWD | **J.W. Davis Company**<br>3030 Canton St.<br>Dallas, TX 75371-0219<br>214-651-7341 Phone<br>800-388-9106 Fax | Manufacturer and distributor of loudspeaker products, electronics, microphones, installation aids, wire and cable, Pataxial loudspeakers, etc. |
| Radio Design Labs™ | **Radio Design Labs**<br>PO Box 1286<br>Carpinteria, CA 93014<br>800-281-2683 Phone<br>800-289-7338 Fax | Manufacturer of sound system electronic components, including preamps, mixers, processors, attenuators, transformers, "Stick-Ons" and "Rack-Ups." |
| RPG® RPG DIFFUSOR SYSTEMS, INC. | **RPG Diffusor Systems, Inc.**<br>651C Commerce Drive<br>Upper Marlboro, MD 20772<br>301-249-0044 Phone<br>301-249-3912 Fax | Manufacturer and designer of acoustical treatment products, including diffusors, reflectors, absorbers, diffusor-blocks, etc. |
| rh RENKUS-HEINZ PROFESSIONAL SOUND PRODUCTS | **Renkus-Heinz, Inc.**<br>17191 Armstrong Ave.<br>Irvine, CA 92714<br>714-250-0166 Phone<br>714-250-1035 Fax | Manufacturer of professional loudspeakers, horns, drivers and processor-controlled loudspeaker systems. |
| SHURE® THE SOUND OF THE PROFESSIONALS... WORLDWIDE | **Shure Bros. Inc.**<br>222 Hartrey Ave.<br>Evanston, IL 60202-3696<br>708-866-2200 Phone<br>708-866-2279 Fax | Manufacturer of microphones, mixers, teleconferencing systems, wireless mics, broadcast electronics, etc. |
| TOA **TOA Electronics** | **TOA Electronics, Inc.**<br>601 Gateway Blvd. Ste. 300<br>South San Francisco, CA 94080<br>415-588-2538 Phone<br>415-588-3349 Fax | Manufacturer of horns, loudspeakers, enclosures, electronics, microphones, wireless, DSP processors, etc. |
| TOA **TOA Corporation** | **TOA Corporation, Inc.**<br>2-1 Takamatsu-cho<br>Takarazuka<br>Hyogo-pref 665 JAPAN<br>0797-71-2211 Phone | Manufacturer of horns, loudspeakers, enclosures, electronics, microphones, wireless, DSP processors, etc. |
| WPW WEST PENN WIRE CORP. | **West Penn Wire/CDT**<br>2833 W. Chestnut St.<br>Washington, PA 15301<br>800-245-4964 Phone<br>412-222-6420 Fax | Manufacturer of wire and cable products, including cables for permanent installation and fiber optics. |

**We are grateful to our sponsors for their ongoing support...**

**Syn-Aud-Con** Synergetic Audio Concepts

# Upcoming Seminars

*Hands-on Training in:*
**Room Evaluation**
**Gain Structure**
**Loudspeaker Arrays**
**Synchronization**
**Equalization**
**and more...**

The following seminars have been scheduled for first half of 1996. Additional 1996 seminars and workshops are pending, and will be added to the schedule as they develop.

Sound System Design Seminar
**February 19-21, 1996**
Atlanta, GA

Sound System Design Seminar
**March 25-27, 1996**
Southern Indiana

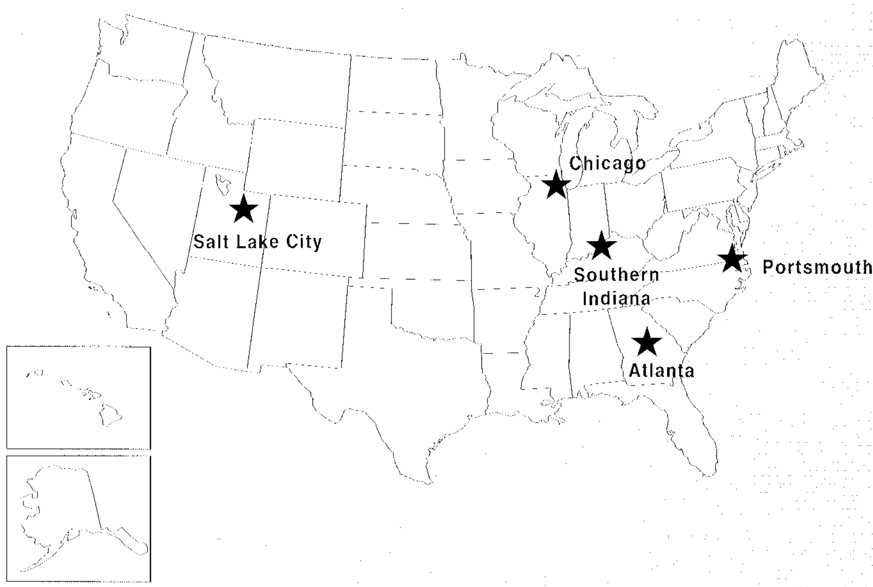
Sound System Operation Seminar
**April 1-2, 1996**
Portsmouth, VA

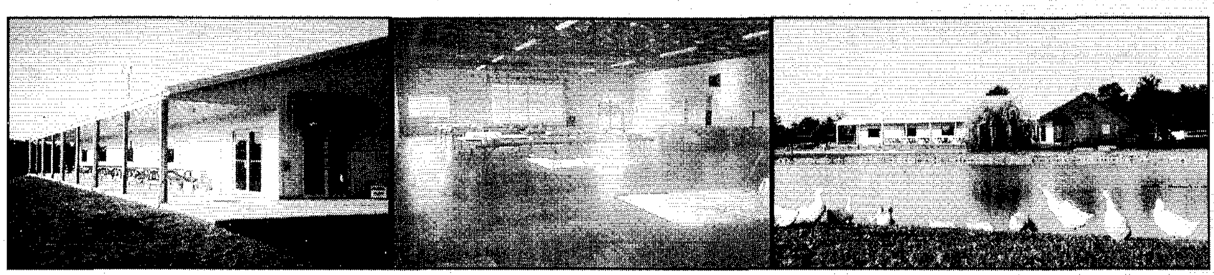
Sound System Design Seminar
**April 14-16, 1996**
Chicago, IL

Advanced Sound System Design
**April 29-May 1, 1996**
Southern Indiana

Sound System Design Seminar
**June 10-12, 1996**
Salt Lake City, UT

Sound System Operation Seminar
**July 22-23, 1996**
Southern Indiana



*The rural setting of our Indiana seminars make them a favorite for attendees*