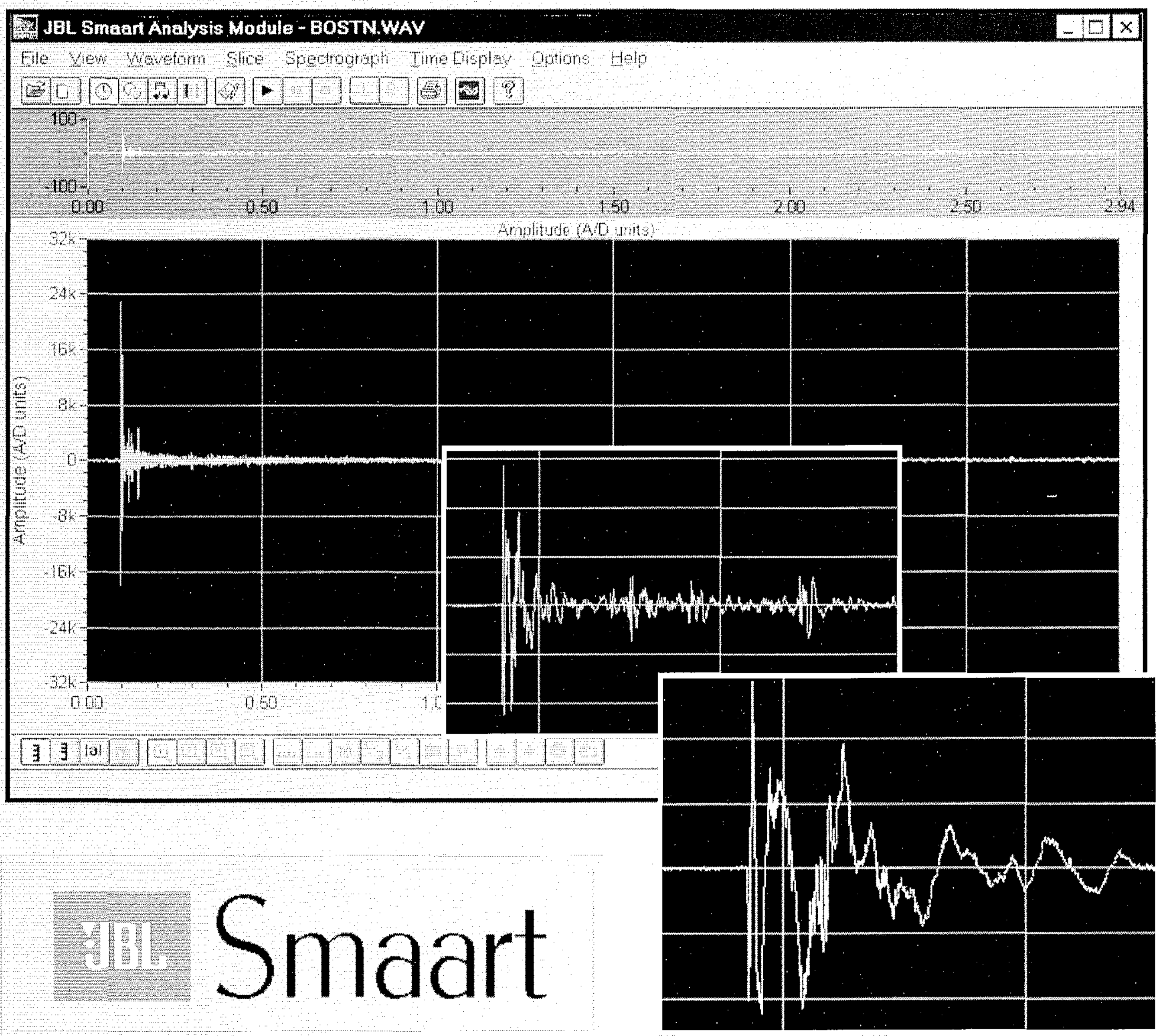
SYNERGETIC AUDIO CONCEPTS
SYN AUD CON

newsletter

Volume 24, Number 2
Spring 1996

JBL Announces Measurement and Analysis Software



JBL Smaart

SYNERGETIC SYN AUD CON AUDIO CONCEPTS

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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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Special Supplement to Newsletter Vol. 24, No. 2

Tech Topic "Digital Basics"

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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A New Tool From an Old Pro

Syn-Aud-Con has always believed that contractors and consultants are seriously impaired without the ability to make meaningful precision acoustical and electrical measurements. The TEF analyzer has brought audio and acoustical measurement out of the dark ages, and has made precision system measurement an integral part of any major project. Measurement systems have progressed tremendously in recent years, requiring the contractor/consultant to either ignore it altogether (as many do) or strive to "keep up" with the furious pace of technology.

It has always pained me to see contractors and consultants avoid measurements completely due to the cost and education involved in acquiring a state-of-the-art measurement system. Perhaps these individuals now have an entrance into the world of system measurement with the introduction of JBL-Smaart, a software-based measurement package running under Windows. It is our hope that this tool will help those who are not measuring to begin to do so. The result of better equipped designers and installers will be better sounding systems.

Pat Brown

JBL-Smaart was developed to provide sound system contractors, acoustical consultants and audio professionals with an accurate and easy-to-use sound system optimization and acoustical analysis tool.

As a longtime sponsor of Syn-Aud-Con, JBL Professional recognized the need for a set of practical measurement tools designed specifically for the sound contracting and acoustical consulting community. The recent increase in audience awareness and expectations for sound quality means that it is more critical than ever for sound systems to reach their highest potential in terms of performance. JBL-SMAART contains both real-time and disk based analysis tools, which allow users to adjust equalizer and delay settings quickly and accurately, using a Windows based computer equipped with a standard Windows sound card. Recent advances in both computer power and sound card technology allow these tools to be implemented with a user interface designed for use in the real-world.

The JBL-Smaart Real-Time module contains many of the features of a standard FFT-based spectrum analyzer enhanced by transfer function capability. The transfer function provides a frequency domain comparison of two signals. This comparison is an extremely powerful tool when comparing the input and output of either an entire system or a single system component. The transfer function feature is enhanced by a Delay Locator, used to measure delay times for a distributed system or signal alignment for transfer function measurements. A coherence function is also included to help determine the quality of transfer function data. The ability to store, display and compare multiple transfer function results is the basis for a powerful process of sound system optimization.

At this point it is critical to note that like any tool, JBL-SMAART will NOT allow a user to turn a poor sound system design into a successful sound system. Rather, using the tools contained within the JBL-SMAART package, a sound system can be optimized to its fullest potential.

The Analysis Module within JBL-Smaart provides disk based analysis of signals collected with either the Real-Time Module Delay Locator, or any other standard Windows wave file. In addition to the standard FFT display, the Analysis Module provides 3D-color spectrographs, time domain plots and Time Slices™. The Time Slice feature allows magnitude vs. time plots of a given frequency range in narrow-band, 1/3 or 1/1 octave resolution, allowing decay rates and reflection structures to be clearly identified.

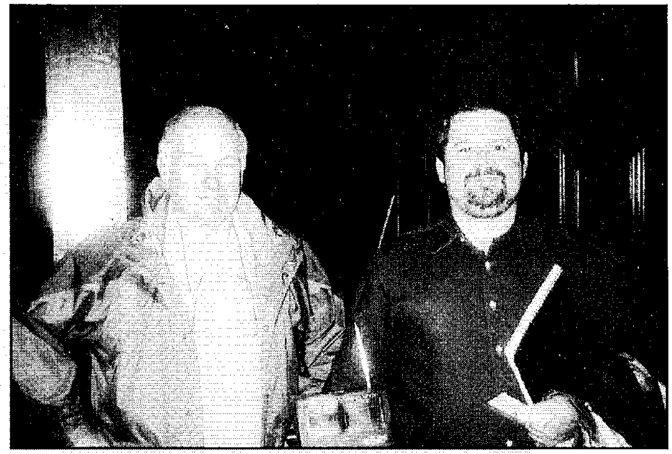
JBL-Smaart was developed by Sam Berkow, an acoustical consultant with a deep interest in acoustic measurement, and well-known sound system designer Alexander Yuill-Thornton. As part of the development process, JBL-SMAART has been used to equalize sound systems in 4 NBA arenas, several amphitheaters, concert halls, recording studios and several major television award show broadcasts.

By using the Windows interface and sound card technology, JBL-Smaart does not require dedicated DSP hardware. What is required is a computer with sound card and an appropriate mic preamp to interface to the outside world. Priced at $695.00, JBL-Smaart is an affordable and practical tool for the working sound contractor or acoustical consultant. JBL-SMAART will be formally introduced simultaneously at the NSCA show in St. Louis and AES Europe in Copenhagen. JBL-SMAART will be available from your JBL sales representative or JBL distributor internationally.

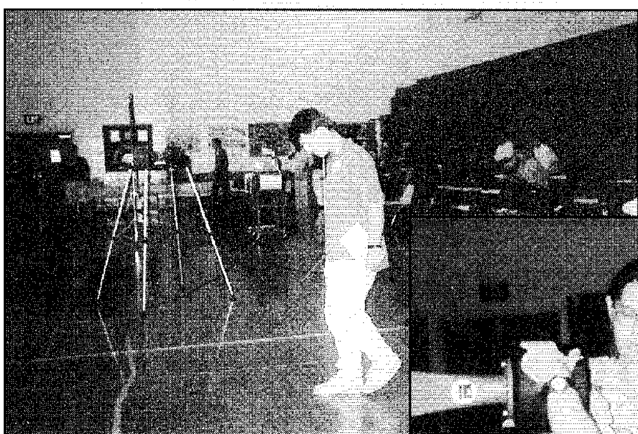
Sam Berkow
Smaart Developer

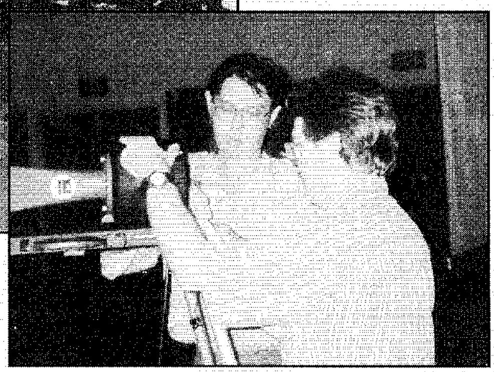
President and Son

Goldline is known throughout the industry for their innovative, affordable and useful product line. We were pleased to spend time with company president Martin Miller and son, Greg during our Fall River Seminar last year. Greg attended the seminar and demonstrated his DSP30 real-time analyzer to the attendees. We were pleased to see the company introduce a full-featured portable impedance meter to the marketplace last year, and look forward to future offerings from this New England company.



Martin and Greg Miller





Right: Randy Bauske and Richard Recktenwald adjust a horn during a class demo.

Moonwalk, anyone?

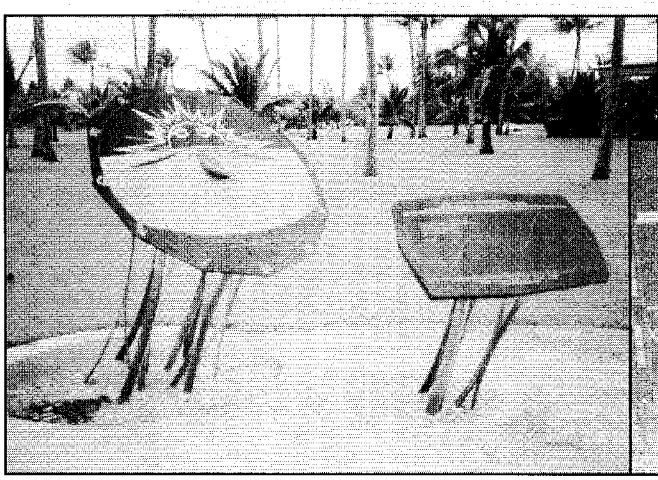
No, we don't teach the moonwalk at Syn-Aud-Con seminars. That's just Joe Antancio stepping off a room dimension during our Hawaii class. Joe works for Baus Engineering, who provided some much needed equipment for the class. In fact, company president Randy Bauske took time out from his busy schedule to attend the class, and offered some insights that come from extensive field experience.

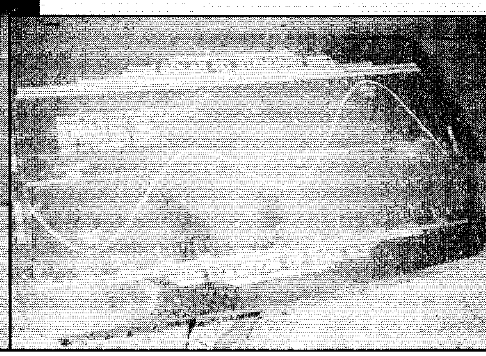
We look forward to a return to the Islands and the Polynesian Cultural Center for a Sound System Operation seminar in 1996.

Those Cycles Keep Coming Around

I was walking to the class at BYU University on Oahu when a sundial caught my eye. Of special interest was the carving on the right that relates solar time to standard. Much of what we do in audio is follows principles found elsewhere in nature, and judging from the sinusoidal nature of the carving, this is but one more case.

One sometimes wonders at the simplicity and complexity of our universe. Perhaps the answer to what lies "out there" can be found right hear.





A clarification on a previous article...

Dear Pat,

I want to respond to an article in your NEWSLETTER Vol 23, No. 1, Fall 1995. On page 7 Don writes about two meetings with Mr. Hamm. At this time I was also present during the visit of Don and Carolyn to the Academy of Art in the former East Berlin.

In the morning of this day we all have had in West Berlin a seminar with Don at Jansen Light and Sound and Mr. Primbs of this company had made the proposal to go to the East and also to visit me at home. But to get the permission it was neccessary to have an "official" purpose.

At this time Jansen Light and Sound had installed the new sound system in the mentioned Academy and we decided to use this hall as our official reason. This way Don and Carolyn met Mr. Hamm. He wasn't the host of this tour but was an employee there, the sound engineer. The tour wasn't video taped in the sense of a security service "Stasi" or so, but another employee of the sound crew has made some recordings of the world famous sound consultant Don and his wife for his private purpose. I have seen this tape later. I didn't seen him making recordings, otherwise I would have asked him to ask Don for permission for these recordings. I apologize that Don and Carolyn have had this distress since that time. But maybe the time of the cold war has a little bit increased the sensation to see behind every normal actions activities of the KGB or his East German partner. This was the same with this exibition dedicated to Angela Davis. As I now found out, at this time every week or twothe topic of the exhibition was changed, and accidently during the visit of Don and Carolyn was the Angela Davis show. The former chief of Mr. Hamm (now retired) had the idea to show the exibition because for his understanding he wanted to show we have also "knowledge of American problems". For a communist leader this was normal at this time and we have had to live with these "demonstrations" every day. But this has had nothing to do with Mr. Hamm, who at this time had organized a small snack with sandwiches and tea as far as I remember.

After the wall came down Mr. Hamm lost his job because the East Berlin Academy of Art was closed and Mr. Hamm had worked two years with Mr. Primbs. 1992 he founded the PROCOM company and because of my recomendation he came together with Mr. Mueller, the new TEF representative in Germany. They have since this time sold more than 20 devices after 3 years not selling one. I assure you that Mr. Hamm is a honest man and doesn't deserve a bad reputation. We are very sensative in Germany to such messages like in your Newsletter and this could cost him and his partner the job and the clients for TEF.

Thank you for your understanding,
Best recards also to Don and Carolyn
Wolfgang Ahnert
Berlin, 27-2-96

What's in a Name?

We get lots of variations on the business name. Some recent ones include:

Synergistic Audio
Audio Synergy Con
Syn-Aud-Com
Cybernetic Audio

I'm sure that Don and Carolyn have many more, having travelled and taught for 23 years. For the record, it's **Synergetic Audio Concepts** *or* **Syn-Aud-Con.**

The Atlanta '96 Class

Scenes from the LIVE SOUND REINFORCEMENT WORKSHOP

The 1996 Live Sound Reinforcement Workshop was a big success, with an attendance of 99 audio people from all parts of the globe. In addition to the standard fare, there were excellent special sessions by Dr. Floyd Toole, PhD., Steve McCale, and John Murray.

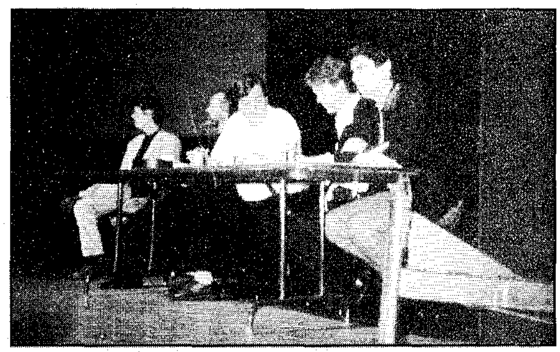
Top Row L-R: David Scheirman, Albert Lecesse, David Robb Steve McCale, Pat Brown Bottom Row L-R: Howard Page, Paul Gallo, Will Parry, Mick Whelan.
Not pictured: Dr. Floyd Toole, John Murray

Dave Revel and Sound Image provided plenty of reinforcement.

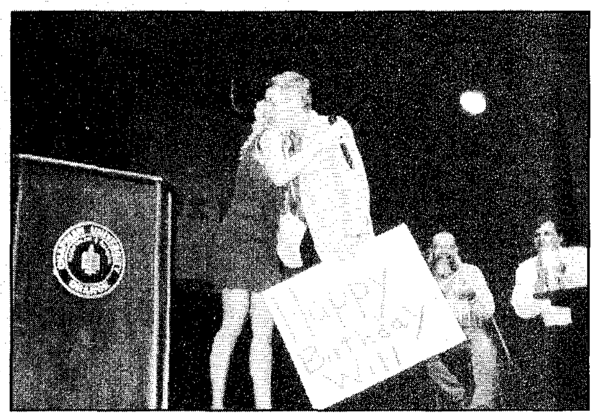
Howard Page shares some of his hard-earned mixing techniques.

Work is already underway on planning the 1997 event, which promises to be bigger and better than ever.

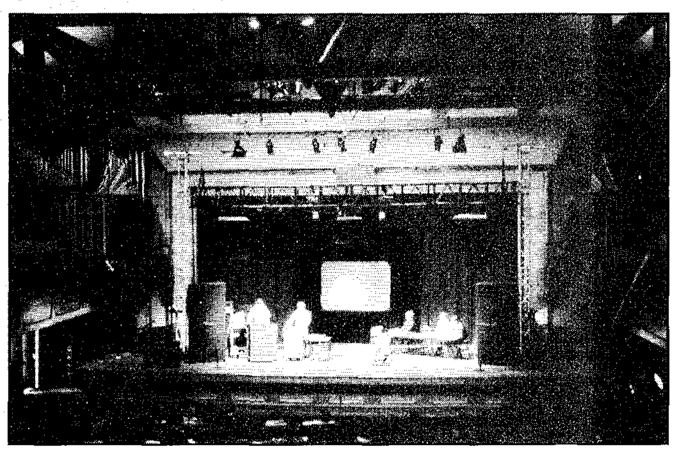
Special thanks to all involved for their hard work, and especially to Paul Gallo and Pro Sound News for their generous sponsorship of the event.



The roundtable sessions provided an opportunity for interesting (and often humorous) interaction with the staff.



Caught in the act! Will Parry's birthday was no secret this year, and a singing telegram brought it to the attention of all.



Chapman University's excellent facilities, when combined with the warm Southern California climate, provided a nice getaway.

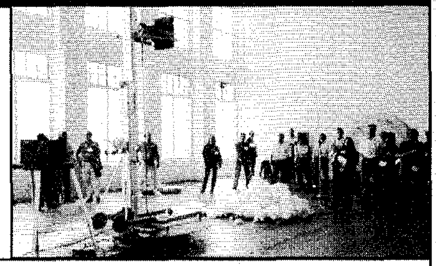
Sound System Training Seminars

Please see the back page for current schedule...

Sound System Operation

Syn-Aud-Con Level One

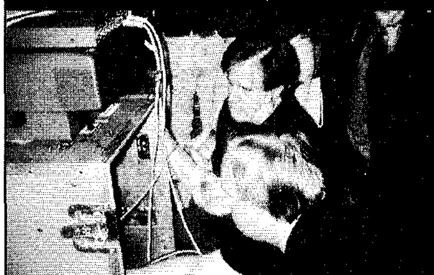
This two-day entry level course is design to familiarize the participant with the basic concepts of sound reinforcements systems. The seminar focuses on the basic workings of a sound system, and how to bring the existing sound system to its maximum potential. If your sound system duties primarily include operating and maintaining an existing system, this seminar will provide you with the knowledge and skills that you need to complete the task.



- Basic Theory
- System Components
  - Microphones
  - Mixers
  - Equalizers
  - Processors
  - Amplifiers
  - Loudspeakers
- Interconnecting Components
- Setting Levels
- Testing Polarity
- Basic Grounding Practices
- Basic Wiring Practices
- System Setup and Operation
- Loudspeaker Basics
- Loudspeaker Placement
- Microphone Placement
- Doing a Sound Check
- Adjusting the Equalizer
- Reducing Feedback
- and more...

Sound System Design

Syn-Aud-Con Level Two



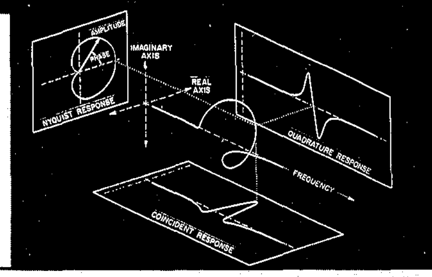
For 23 years, Syn-Aud-Con has been providing training in designing sound reinforcement systems. The Design seminar is the most comprehensive short-course in system design training available today. The contractor, consultant, and advanced system operator will enjoy this three-day audio and acoustic experience, and will leave with knowledge tools necessary to design sound reinforcement systems from the drawing board stage of the project. If system design is your job or interest, then this is your seminar. Level One experience is recommended.

- Design Prerequisites
- The Human Auditory System
- The Decibel in Acoustics
- Evaluating the Acoustic Environment
- Room Acoustics
- Basic Acoustic Measurements
- Speech Intelligibility
- 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
- Acoustic Gain
- Loudspeaker Parameters
- Loudspeaker Arrays
- Distributed Systems
- Calculating Coverage
- Array Synchronization
- Equalization Techniques
- and more...

Advanced System Design

Syn-Aud-Con Level Three

Audio and acoustic technology is constantly evolving, and the Advanced System Design Seminar is designed to keep the professional or serious amateur current with the latest standards and techniques. This three-day seminar places a strong emphasis on the role of the computer as a design, measurement and system operation tool, and is an invaluable aid for the system designer in need of training in this vital area. Level Two experience is recommended.



- Computer Room Models
- Ray Tracing Methods
- Image-Source Methods
- Array Models
- Auralization Techniques
- The Impulse Response
- FFT Measurements
- TDS Measurements
- The Analytic Signal
- The Heyser Spiral
- Measuring Speech Intelligibility
- Noise Measurements
- Array Calibration Techniques
- Computer-Controlled Systems
- Electronic Acoustic Enhancement
- Intelligent Sound Systems
- Multiple Amplifier Control
- Digital Signal Processors
- Advanced Equalization Methods
- Automatic Mixers
- Computer-Controlled Signal Routing
- and more...

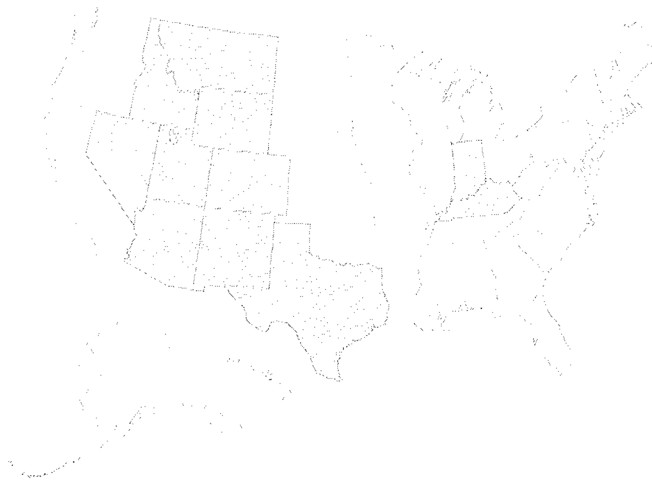
Syn-Aud-Con Appoints Two New Reps

Syn-Aud-Con is pleased to announce the addition of two new rep firms to aid in the promotion and organization of the sound system training seminars. Pro Tech Marketing will cover the West/Southwest and A/V Marketing will cover Indiana and Kentucky.

Our reps play a vital role in spreading the word about upcoming seminars to their territories. We look forward to working with these two fine organizations on future seminars.

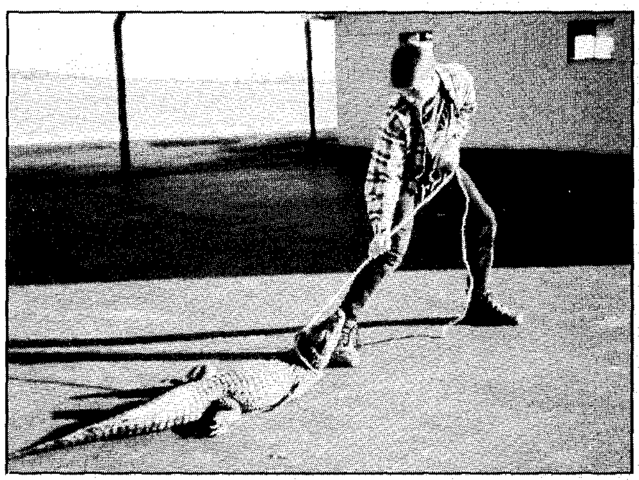
A/V Marketing
1-317-387-7400 Phone
1-317-387-7410 Fax

Pro Tech Marketing
1-801-561-8844 Phone
1-801-561-9969 Fax



Indy Gator?

Alligators are not indigenous to Indiana, but nobody told this one. The six-foot gator belonged to a local resident and escaped to an area farm pond. Imagine baitin' up the hook and pulling this thing in! Pictured is Justin Brown antagonizing the reptile (rather than his sisters, pictured below).



Musical Aspirations...

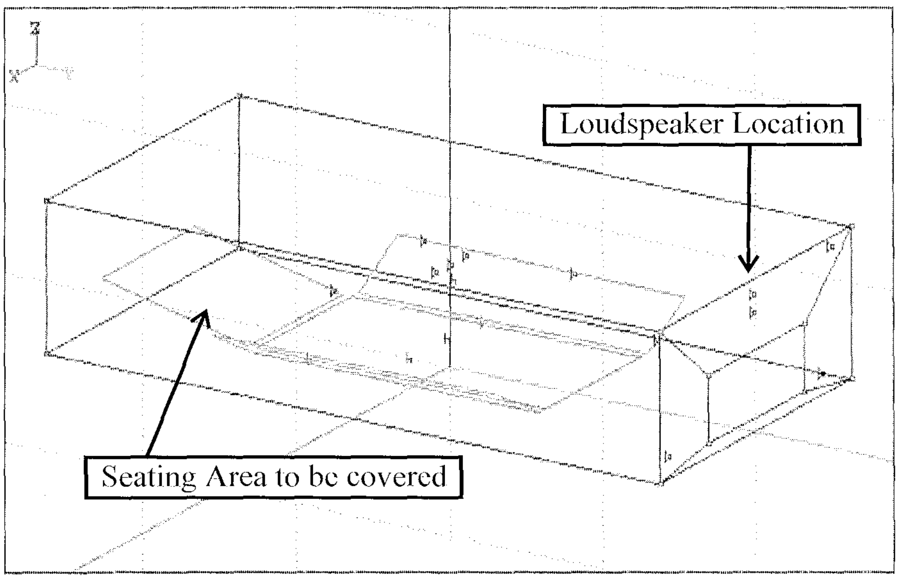
I was beginning to think that none of our offspring would take an interest in the guitar (an activity that consumed much of my early years) until the youngest one, Heather, decided to give it a try. That's her and older sister, Ashley, picking at Dad's (eeek!) guitar. Needless to say she now has her own and looks forward to each new lesson.

A Computer Model Helps Solve a Common Design Problem

There are many factors involved in designing a sound system to provide coverage to an audience. Due to the complexity of the problem, we prefer to supplement our intuition and "horse sense" with the modern tools available. This short study reveals that, in this case, intuition alone might have gotten us into trouble.

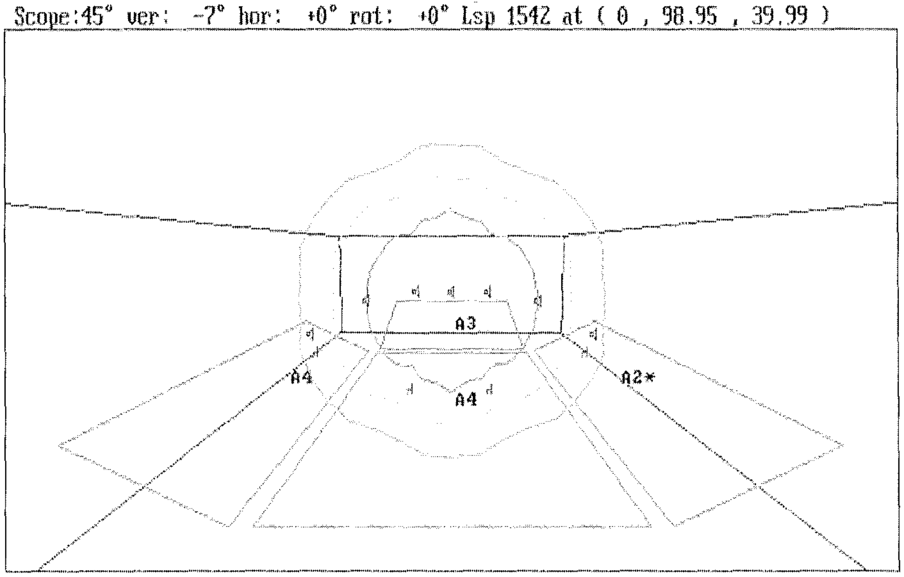
The room pictured was modelled in the EASE sound system design program. It is a fairly common case of a multi-purpose facility that must double as a theater and basketball arena. The criteria for the basketball system is primarily intelligible speech. The theater system has the same the criteria, but with the addition of the desire to have the sound image from the stage area to add realism to the performances.

There are a number of ways to approach the problem, and this study considers two.

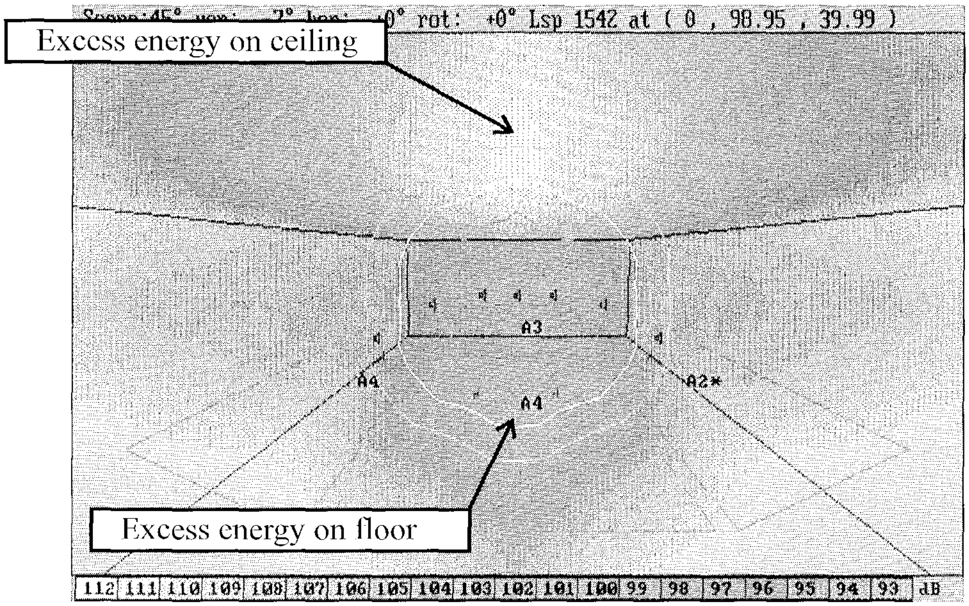


The EASE program will allow us to consider the possible solutions to the problem at the drawing board stage of the project. The advantages are obvious, the primary one being the avoidance of installing an inferior system (at possibly a higher price) that will never live up the expectations of the client. The figure (left) shows the location of the far-throw horn being considered to provide coverage to the remote seating area.

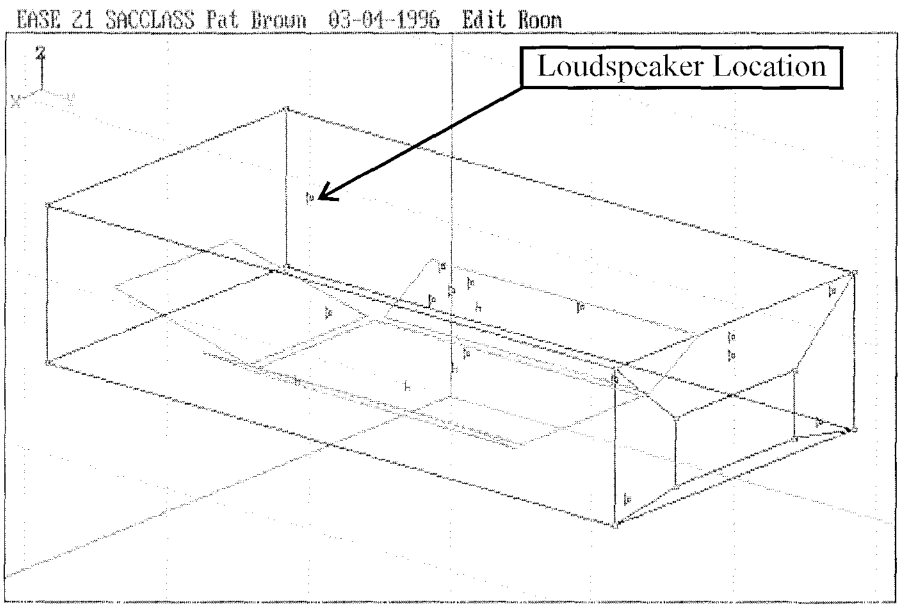
Scope:45° ver: -7° hor: +0° rot: +0° Lsp 1542 at (0 , 98.95 , 39.99)



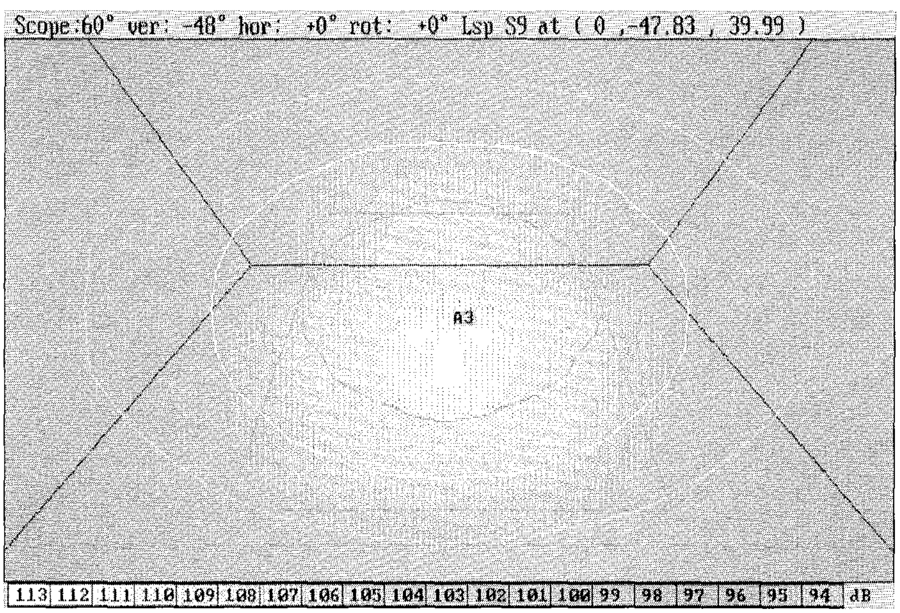
Using the *Speaker View* feature of the EASE design program, it appears that the long-throw horn will provide adequate coverage of the remote area. Almost all of the seats lie within the -3dB contour of the loudspeaker coverage. What is often not considered is that the distance is increasing within the area of coverage, and the effects of inverse-square law (the spreading of sound energy with distance, resulting in a 6dB drop each time the distance doubles) must be accounted for.



Next, the program can scan the seating area to calculate the coverage. This feature accounts for the loudspeaker coverage and inverse-square law level change with distance. Note that when viewed in this manner, the loudest areas are actually the ceiling and floor, since these are closer to the loudspeaker. The goal is to confine the radiated energy to the seats that need it, and it becomes apparent that this approach is not working very well.



Another possibility is to utilize a satellite loudspeaker to cover the remote area. It can be placed near the area and delayed to place its arrival time about 20 ms behind the sound arriving from the short-throw horn. This will allow the sound to image from the stage area, while actually coming from overhead.



Rescanning the seating area reveals that the sound energy is now well-confined to the seats that need it. As a result, significantly less energy is spilling over and exciting the room. This is accounted for mathematically in system design equations by the Ma factor, which accounts for the fact that confining the energy to the audience area will reduce the amount of energy that excites a reverberant sound field (if present and of significance).

Identifying Sound Fields

More on Quantifying Reflected Energy Within the Auditorium

The first step to designing a sound system for any space is to consider what sound fields might be present in that space. The three main sound fields of interest to the system designer (apart from noise) are the direct field, the early-reflected field, and the reverberant field. They can be defined as follows:

The Direct Sound Field

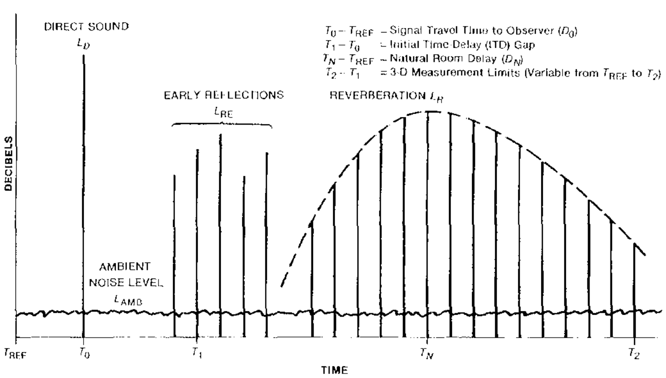
The direct sound field is the energy that arrives at the listeners ear without having encountered the space. Its travel path is usually straight from the loudspeaker to the listener. The direct sound field has some interesting characteristics in that it is not room dependent.

The Early-Reflected Sound Field

The early-reflected sound field consists of discrete energy arrivals from room surfaces. It always arrives after the direct sound field, and in most auditoriums continues for about 200 ms. It is this sound field that is usually being considered when one describes the sound of the room as "hard" "soft" "live" etc. The early-reflected field is quite useful to the listener, providing support and texture to the direct sound field.

The Reverberant Sound Field

The reverberant sound field is actually the accumulation of energy in the space, and is typically defined as the time it takes for the accumulated energy to decay by 60 dB after the



cessation of a steady sound source. An analogy might be useful in understand what reverberation is.

Consider a broken pipe in your basement. Near the break the water is spraying out in distinct streams and under great pressure. If the break were to go unfixed, there would be a uniform accumulation of water on the floor throughout the basement. The leak is manifesting itself in at least two ways, one

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

English Units

where V is the room volume in feet or meters
S is the total surface area in feet or meters
a is the average absorption coefficient
c is the speed of sound

$$RT_{60} = \frac{55.26V}{Sac}$$

SI Units

The classic Sabine equation

being the spray and the other the accumulation. A reverberation measurement is analogous to measuring the depth of the water on the floor, at a point remote from the broken pipe. A yardstick would be useful for checking the depth of water on the floor, but would be pretty useless in evaluating the conditions surrounding the actual break. It quantifies the accumulation of energy (water) that results from the rooms ability to retain it. Many rooms do not accumulate significant energy in this manner, which makes the use of statistical equations and system parameters unnecessary in such spaces.

The Sabine Equation is the most useful for an initial evaluation of the space, primarily due to its simplicity. When the volume of the room is known, the RT_{60} is simply a function of the absorption present. It must be remembered that the Sabine equation assumes that the absorption is uniformly distributed throughout the space, such as a marble-laden cathedral. Even if this is not the case, such as a church with carpet and/or pew cushions, it is prudent to consider what the RT_{60} would be without these in place.

The Fitzroy Equation

When the absorption is primarily on one or two major room surfaces, such as the floor or ceiling, the statistical validity of the Sabine equation is weakened. The tool of choice becomes the Fitzroy equation, since it allows the effect of each surface to be considered independently.

$$RT_{60} = \frac{X}{S}\left[\frac{0.049V}{S\bar{a}_X}\right] + \frac{Y}{S}\left[\frac{0.049V}{S\bar{a}_Y}\right] + \frac{Z}{S}\left[\frac{0.049V}{S\bar{a}_Z}\right]$$

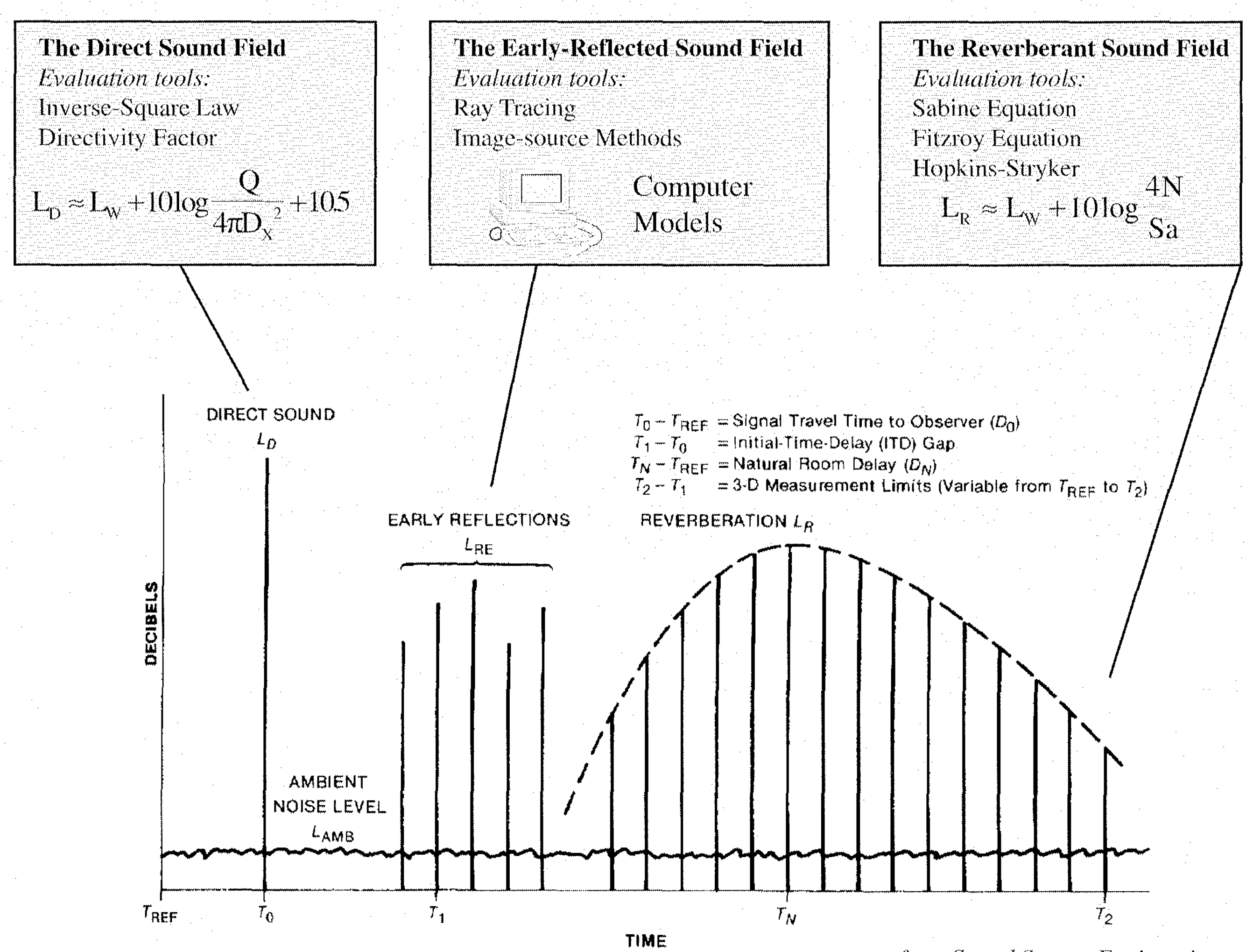
X is the total surface area of the parallel surfaces on the X axis.
Y is the total surface area of the parallel surfaces on the Y axis.
Z is the total surface area of the parallel surfaces on the Z axis.

The Fitzroy equation

The early energy arrivals are of prime importance, and constitute the most significant sound field other than the direct in most spaces. The non-statistical nature of this sound field (varies from seat to seat) makes statistical equations useless in evaluating it. At the drawing board stage of the system design, room modelling becomes our most valuable tool in estimating the time interval, level and number of reflections. Opponents of computer modelling programs often site that the computer must make many assumptions when modelling rooms, but not as many as must be made with a protractor, straight-edge and human intuition. While the computer model isn't perfect, it is very useful for estimating the effects of inverse-square law, absorption, room geometry and loudspeaker directivity on the performance of the system.

Statistical reverberation equations are intended for use in spaces with significant statistical reverberation, which rules out studio control rooms, most class rooms, and many churches and theaters.

Drawing Board Tools for Evaluating Rooms



from Sound System Engineering

Cycles, Circles and Triangles

Part II - The Impulse Response

Last issue we looked at some fundamental concepts concerning audio and acoustic signals, which included some of the various data displays commonly associated with such measurements. Such an overview is useful for getting a feel for the subject, and affords us the opportunity to "zoom in" on some specifics.

The Impulse Response

The impulse response is perhaps the most fundamental of all audio and acoustic measurements. In fact, it is the cornerstone of much of what we do in modern signal analysis.

The impulse response of a system describes the time domain behavior of that system to a well-behaved input. While there are a variety of techniques in existence for "pulsing" a system, the most fundamental stimulus consists of a very short burst of energy (hence the term "impulse" response). The perfect impulse would be infinitely short in duration, and would have no rise time or decay time. This "theoretical" model can only be approximated in real life. Once input into the system, the response of the system is then recorded and analyzed. Even though the same impulse were fed to a variety of systems (mixers, equalizers, loudspeakers, etc.), the response of each of these systems to the impulse would be unique. A good example from everyday life is a drum kit. The drummer uses the same stick (impulse) to strike each drum, but each has its own unique sound. An acoustic piano is another good example where one type of impulse can cause many types of responses.

The impulse can be thought of as an electrical drum stick. We "strike" the system under test and record and analyze the result.

Once the time response of the system is obtained, this can be displayed as a frequency response of the system. The mathematical "map" between the time and frequency domains is the Fourier Transform. It can be thought of as a mathematical filter which can take time information as its input and convert it to frequency domain information. This is possible because time and frequency are actually two different ways of describing the same event. For instance, a time domain description of a bus schedule would be that the bus comes at 1, 2, 3, 4 and 5 o'clock. A frequency domain description of the same event would be that the bus comes every hour beginning at one o'clock. Both are valid descriptions of the bus schedule. As such, we can actually refer to the frequency response as a 1/T response, where T stands for time. FFT analyzers derive their name from the process of taking time domain data and converting it to 1/T (frequency) domain data using the Fast Fourier Transform.

Figure 1 shows an electrical impulse which might be used for a system test. Notice that the impulse is very short, with a very short rise time. Figure 2 is the 1/T or frequency domain view of the impulse. Notice that it has a very wide passband. It becomes apparent that our very short time domain impulse con-

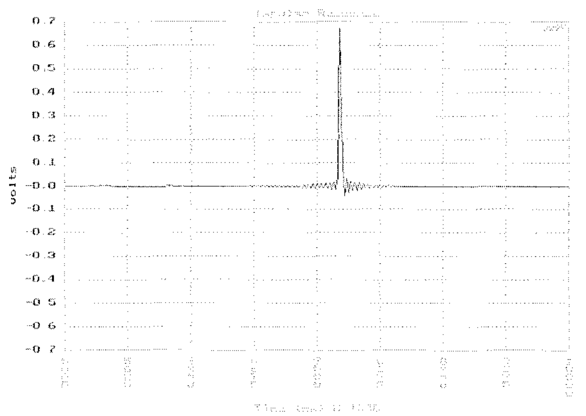


Figure 1 - An impulse

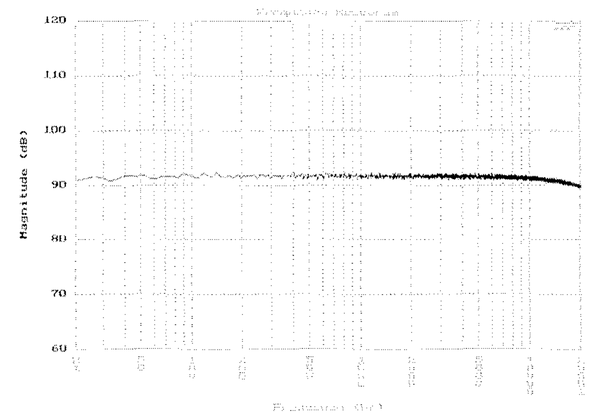


Figure 2 - From a frequency viewpoint

tains all of the frequencies that might exist in our audio system under test. Let us now dissect the impulse and see why this is true.

Figure 3 is the same impulse, but it has been filtered with a one-octave 1 kHz bandpass filter.

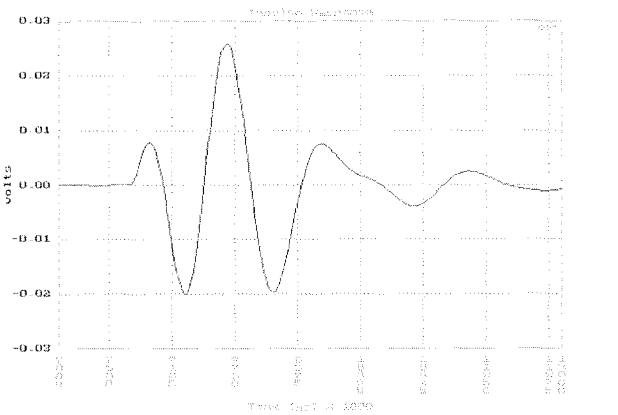


Figure 3

The filter reveals a sinusoidal component in the impulse, whose period is about 1 ms. Viewed in the frequency domain (Figure 4) it appears as a band of energy centered at 1 kHz. It should become apparent that if the procedure were repeated for each octave band, the frequency domain view would have so many bands that it would look like Figure 2.

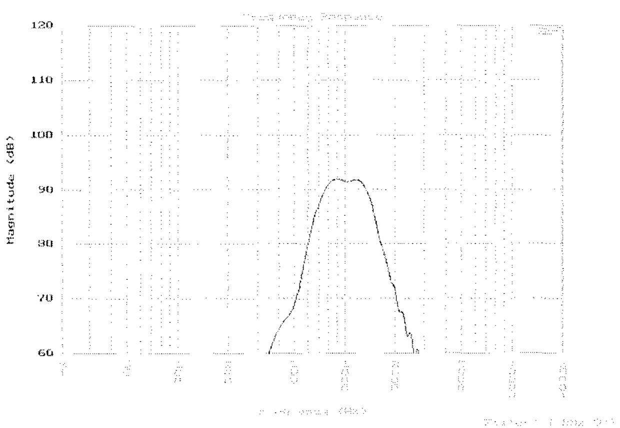


Figure 4

This illustrates that once the impulse response of a system is obtained, there are many possibilities for dissecting and viewing the data. As such, the impulse response of a system completely characterizes that system.

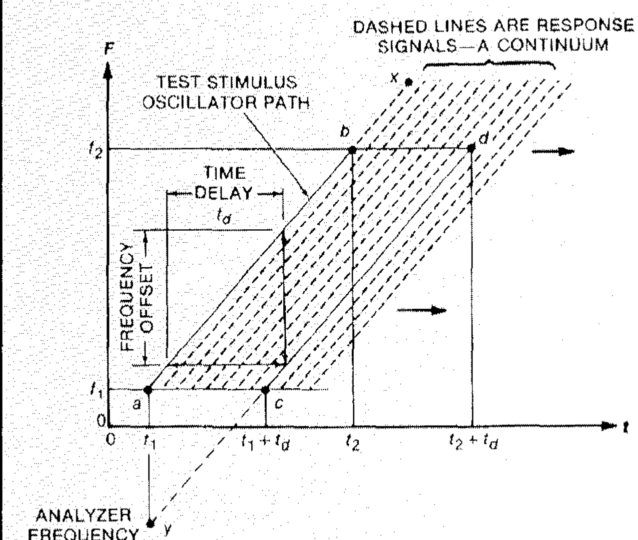
A practical problem that arises is how to get the impulse into the system. The perfect impulse described earlier can make a system behave in a non-linear manner, since it is impossible for a loudspeaker to reproduce such an impulse (and it may destroy itself trying). Such bursts of energy can invoke nonlinear responses in the system under test. So how does one mea-

Time-Delay Spectrometry - A Unique Way of Acquiring an Impulse Response

The impulse-response is a time domain measurement, but that does not mean that the information must be gathered in the time domain. Since the Fourier Transform provides a map between the time and frequency domains, the option exists to perform a frequency domain measurement and display the information as though it were gathered in the time domain. The technique was developed for audio by Dick Heyser of the Jet Propulsion Laboratories in the late seventies.

Instead of using a broad band test signal, such as an impulse or MLS stimulus, Time-Delay Spectrometry (TDS) uses a swept sine wave. The start and stop frequencies are selected to be appropriate for the system under test and the information desired. This is a two-port measurement technique, which means that the analyzer provides the stimulus to the system and then reacquires it for analysis. Since all causal systems involve signal delay between input and output, the receive portion of the analyzer can be offset in time from the send portion to account for this delay, which can be fairly long for large auditoriums. The signal is "re-acquired" from the environment by digitally sampling the output of a tracking filter that is synchronized with the send signal. In effect, the analyzer is only listening for what it sent, and will be less sensitive to all other signals, such as ambient noise and reflections.

The advantage of TDS is superior signal-to-noise ratios than other techniques, allowing meaningful measurements to be made under non-ideal circumstances.



from Sound System Engineering

sure the impulse response of a system without disrupting the system being measured? Violinists have long known that the same note can be sounded by plucking the string or bowing the string. Of course, bowing takes the stimulus and spreads it out over time. Given the proper processing, this same technique can be used to stimulate an audio system. A TDS measurement (Time-Delay Spectrometry) utilizes a swept sine wave in the frequency domain to stimulate the system. The signal is "reacquired" from the environment with a microphone and tracking filter. Even though the measurement is being performed in the frequency domain, the inverse FFT provides a map back to the time domain to display the information as an impulse response. The major advantage of this technique is that a great deal more energy is received by the analyzer, since it is spread out over time rather that a one-shot impulse. Another advantage is that the receive filter can be made frequency selective, meaning that the analyzer will only listen for the signal that it sent. Such techniques offer signal-to-noise benefits that far exceed conventional FFT analyzers. The drawback is that it becomes difficult to obtain large time windows with this technique, making it more applicable for transducer measurements than room measurements.

Most 2-channel FFT analyzers used for audio measurements compare the received signal with the one sent. This comparison yields the transfer function of the device(s) being measured. Transfer function measurements theoretically allow any test stimulus to be used, so long as its bandpass is equal to or greater than the system being measured. This presents some interesting possibilities, such as using the performer as the test stimulus, and observing the loudspeaker response during an actual performance. The drawback is that signal-to-noise is moderate to poor, meaning that typically many measurements must be taken and the results averaged to obtain reliable data.

The system impulse response is typically displayed as a

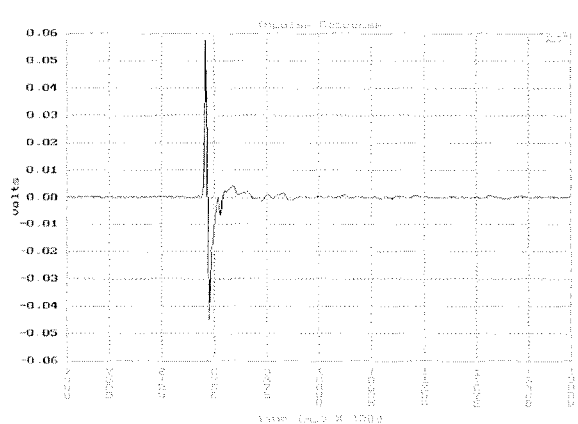


Figure 5 - An impulse response measurement of a small loudspeaker.

pressure vs. time display (see figure 5). The received energy averages about 0, which corresponds to the ambient atmospheric pressure, the resting position of a loudspeaker cone and/or test microphone diaphragm. The polarity of a transducer can be observed from this information by noting whether the application of the impulse to the system results in an over-pressure or under-pressure at the microphone. What we are really seeing in figure 5 is the movement of the measurement microphone diaphragm in response to the fluctuations in sound pressure caused by the loudspeakers movement. This pressure response represents the potential energy component of the loudspeakers energy, also known as the real part. The magnitude response of the system displays the total energy present, which is the complex combination of potential and kinetic energy (See Fig. 6). For linear systems the kinetic energy component can be obtained by several methods. It is possible to measure it directly, (most accurate) or to estimate it by phase shifting the potential component by 90 degrees. Several mathematical processes exist to accomplish this task.

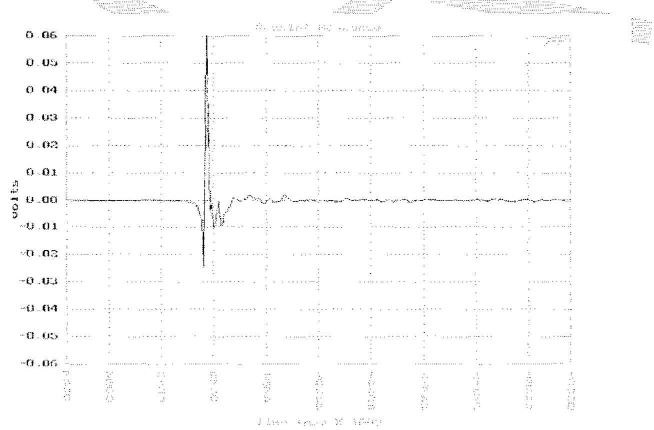
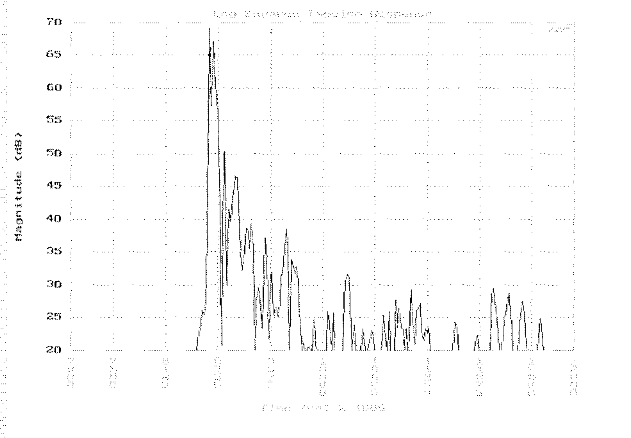


Figure 6 - The "imaginary" component is sometimes derived from the (measured) "real" component.

Once the potential and kinetic parts of the energy are known, a number of displays become possible. Some of the more popular ones are shown on the next page. These are all simply different views of the information yielded by the impulse response of the system.

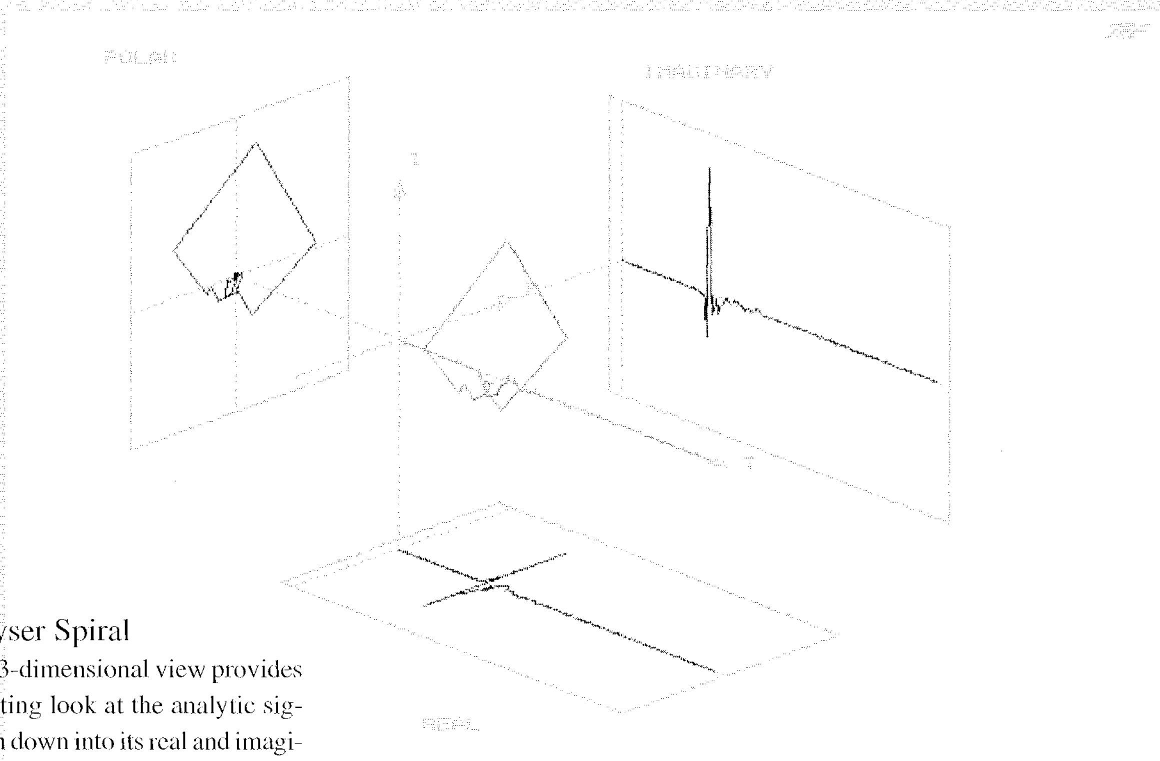
Once the impulse response of a system is obtained, the possibility exists to combine (convolve) it with another impulse response, producing a mixture of the two. The first response may be the measured impulse response of a room, and the second may be dry program material. Such a combination would allow the program material to be "played back" through the room without actually doing so. Methods now exist and are continually evolving for calculating the impulse response of an as yet unbuilt room using a computer model. In effect, the room can be auditioned prior to construction.

Some Various Ways to Display an Impulse Response



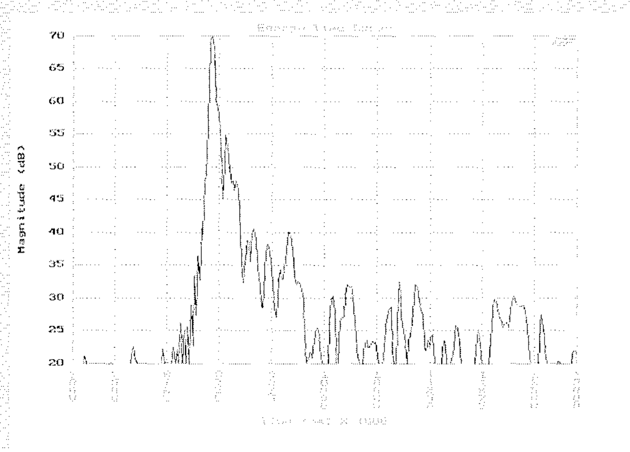
The Log-Squared Response

The log-squared response simply takes the sound pressure (impulse response) and squares it, making all points positive. This information is converted into dB and displayed on a log scale.



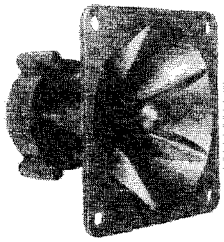
The Heyser Spiral

This 3-dimensional view provides an interesting look at the analytic signal broken down into its real and imaginary parts.



The Energy-Time Curve

The ETC displays the complex combination of the potential (real) and kinetic (imaginary) energies. It represents the total energy at work in the loudspeaker cone or measurement microphone diaphragm.



Piezo Magic

Accurate Gain Structure Using a Common Device

In order to set the gain structure of a signal processing chain properly, it is necessary to determine when each device (mixer, eq, etc.) can no longer provide an increase in output signal with an increase in input signal. This condition, known as clipping, indicates that the limits of voltage swing for that device have been reached. Once determined, a voltage measurement can be made to determine the E_{out} for the device. Traditionally, this task has been performed with an oscilloscope, an instrument that displays voltage vs. time on its screen. E_{out} is easily determined by inputting a sine wave into the system (unloaded) and observing the waveform on the scope (Fig. 1). As the input is increased, the waveform will flatten out when the maximum E_{out} is reached (Fig. 2).

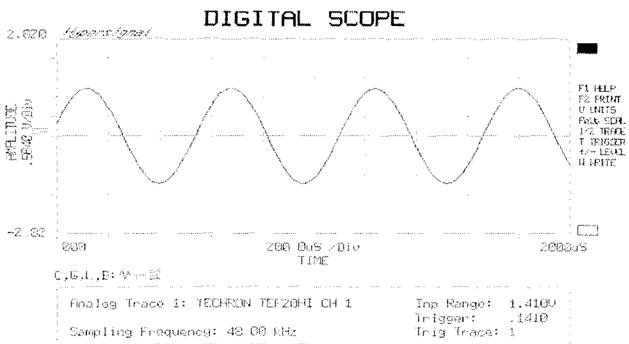


Figure 1

It is also interesting to observe this phenomenon in the frequency domain, where the clipped signal produces harmonics that were not a part of the original test tone. This explains why clipping is such an audible effect, and why it should be avoided (Fig. 3 and 4).

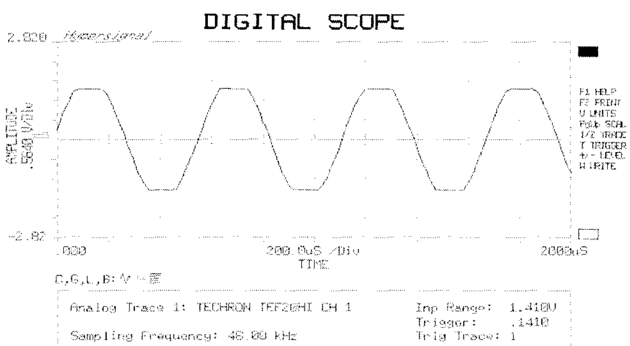


Figure 2

The audible artifacts of clipping provide us with another, much less expensive, method to determine E_{out}. The solution presents itself when the problem is described in detail. In short, a method is needed to monitor the output of a line level device for the presence of the odd harmonic artifacts of clipping. The monitoring device should appear as an open circuit (high impedance) to the device under test at the fundamental frequency used as the system input. A further benefit would be that the monitoring device should be accurate, inexpensive, and readily available. Better yet, if the device were passive, it would require no external power source to complete its task.

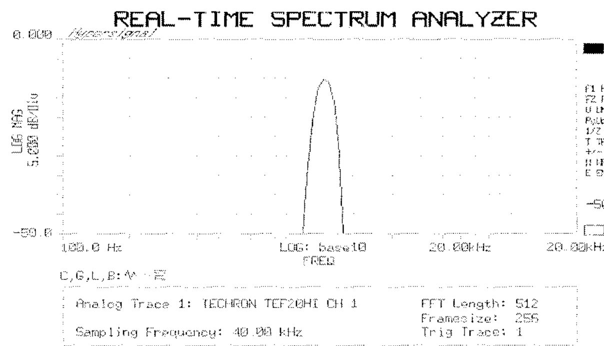


Figure 3

All of these requirements are fulfilled nicely by the common piezo tweeter. This voltage-sensitive high-frequency device is excellent for reproducing the harmonic by-products of clipping. The method I use is to select a test frequency below the passband of the piezo tweeter. This ensures that the impedance will be high (inversely proportional to frequency) and that the fundamental (400 Hz works well) is not audible during the test. As the input signal is increased, the piezo will sound out loudly when the maximum E_{out} (and the resultant clipping) is reached. Bridging a voltmeter across the output at this point will yield E_{out}.

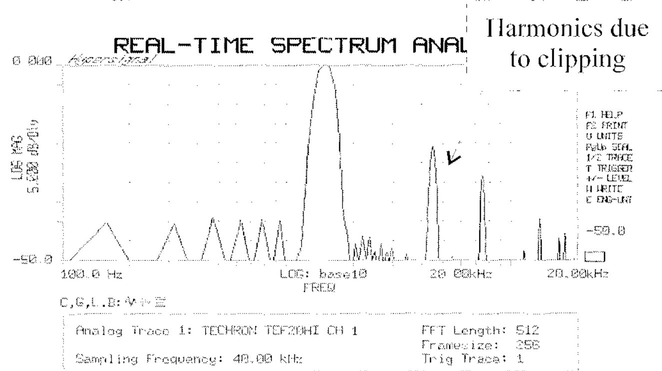


Figure 4

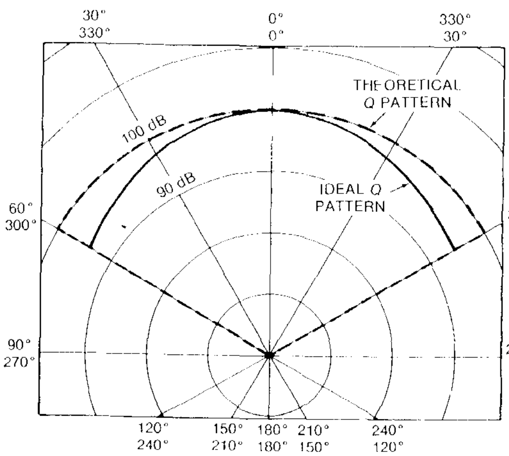
For quick system calibrations, it may be unnecessary to actually measure E_{out}. Simply bring the mixer to clipping, move the device to output of the equalizer and bring it to clipping, and proceed to the output of the system in this manner. Once completed, the needed system headroom can be set with the trim control of the mixer.

Looking Deeper than the Specification Sheet

Sound System Engineering has an excellent appendix entitled "Establishing a Loudspeaker Directivity Figure of Merit." It deals with the fact that there is sometimes a poor correlation between the coverage angles of a loudspeaker and its Directivity Factor, or Q.

Theoretical Q

The well-known Malloy equation calculates the theoretical Q possible if all of the sound energy passed only through the coverage angles and had 0 dB variation over the area bounded by the coverage angles at the surface of a sphere intersecting the angles. The ideal loudspeaker, on the other hand, would put all of the energy into the area bounded by the coverage angles, but would have a level that changed with increasing angle from 0 dB on-axis to - 6 dB at each boundary.



Q of an ideal 120° x 120° loudspeaker = 7.79*
Theoretical Q = 3.7
Suggested multiplier = 2.25 (3.7 x 2.25 = 8.3)

*Calculated by average weighting method every 10°

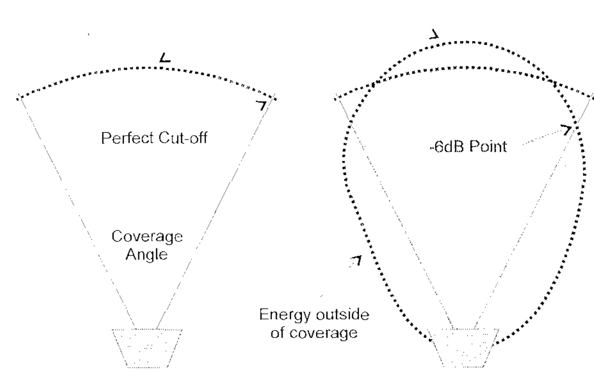
We have stated many times that all speech system designs for reverberant spaces should begin with an evaluation of the acoustic properities of the space, including the RT_{60}. Once in hand, this information can be used to establish the required Q of a device needed to provide an acceptable direct to reverberant ratio for the listener seat under consideration. The Q, in turn, can be turned into a set of angles used to examine the coverage provided by that device.

The problem lies in the fact that all 90 by 40 coverage loudspeakers are not created equal. It is possible (and common) for two devices with the same coverage angles to radiate considerably different amounts of energy outside of their 6 dB down points. This can be quite deceiving when comparing two devices, since the design programs can predict the same coverage from two vastly different devices.

The idea put forth in the afore mentioned appendix is that it is prudent to consider this phenomenon when selecting loudspeakers for a project. For the speech pass band (300 - 5000 Hz) horns will typically exhibit a good correlation between Q and coverage, given that the horn is of sufficient size to provide directivity control at the frequency of interest. Many multi-driver systems correlate very poorly, and if the system designer is considering coverage alone, the results can be dissappointing. It is possible to quantify the correlation between coverage and Q with a loudspeaker Directivity Figure of Merit, which simply involves a comparison of the measured Q with the ideal Q. The ratio can then be expressed as a percentage.

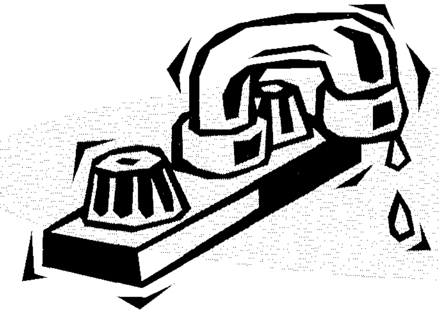
Perhaps one of the most useful applications of a loudspeaker DFM is its application in computer models used for system design. Since few of the design programs consider the L_W of the loudspeaker (looking only at the loudspeaker isobars when considering coverage), loudspeakers with a high DFM should correlate better with the performance of the finished system. It should be pointed out that it is possible for loudspeakers with low DFM's to sound good in some applications. What we are saying here is that it will be difficult to predict the performance of such devices at the drawing board stage of the project, where many disasters can still be averted at minimal cost. The bottom line is that there is more to the game than establishing coverage, and consideration of L_W, Q, DFM, etc. will yield more predictable designs and improved system performance.

$$\mathrm{DFM} = \frac{\text{Measured Q}}{\text{Ideal Q}} \times 100$$



Theoretical Q value
is based on ideal coverage

Real-world Q value
based on device performance



Hot or Cold?

More on Polarity

Last quarter we presented an article on polarity testing in sound reinforcement systems. Maintaining proper polarity is fundamental to good sound system practice, and the ability to check for proper polarity should be fundamental to any system designer/installer. As with anything else in audio, there is more than one way to test polarity. While this is normally a very straight-forward and absolute test in the signal processing stages of the sound system, some transducers do not cooperate, and a different method must be employed.

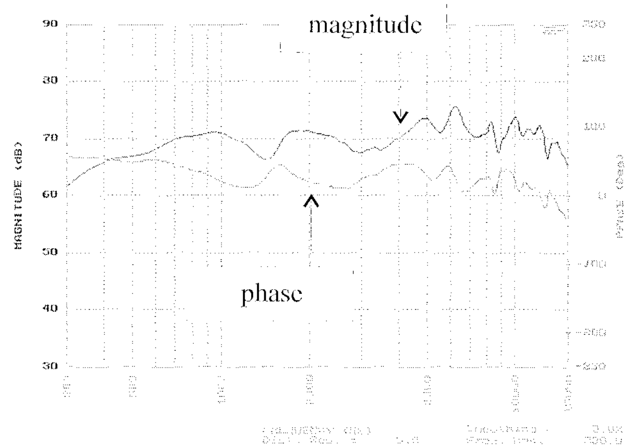


Figure 1

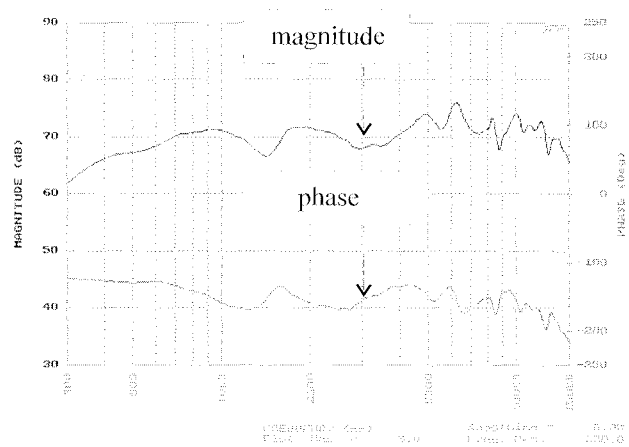


Figure 2

Figure 1 is a magnitude and phase plot of a single-cone loudspeaker. It's polarity is easily determined from the data, as it's phase response averages around zero in the middle of the pass band. Figure 2 is the same device with the loudspeaker leads reversed. Figures 3 and 4 are the Nyquist displays of the same data. The in and out-of-polarity conditions are mirror images of each other. We have always considered this a very good indicator of polarity. Notice that the phase predictably jumps 180 degrees at every frequency, indicating that the polarity has been reversed. All three brands of brands of polarity testers in the lab agreed with the data.

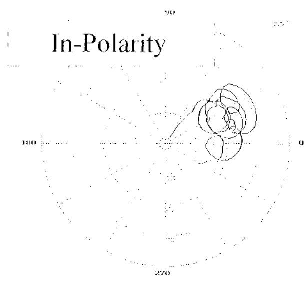


Figure 3

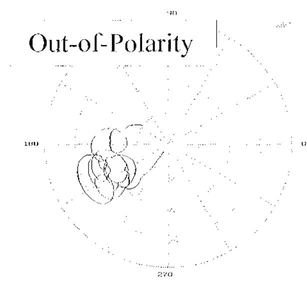
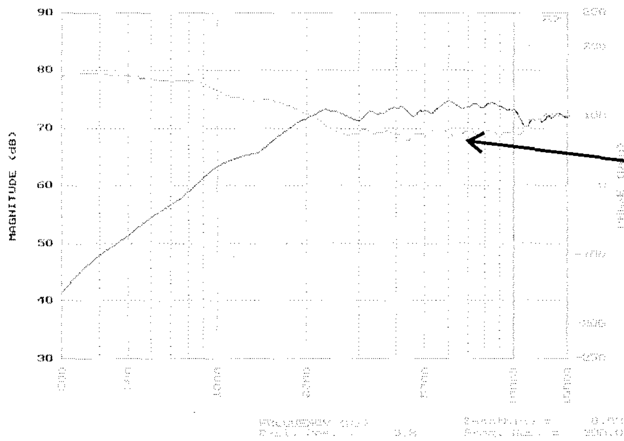


Figure 4

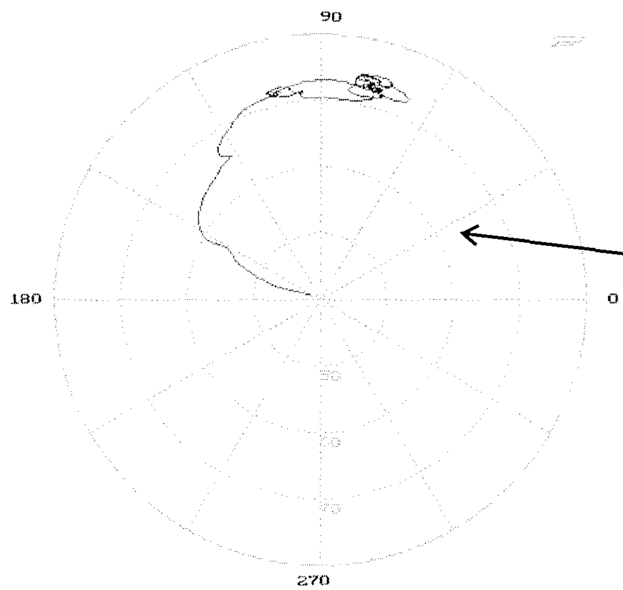
Figure 5 is the magnitude and phase data for a fairly robust dome tweeter. The usual practice of "fine tuning" the receive delay to flatten the slope of the phase results in the phase averaging about 90 degrees. The Nyquist display shows that data is distributed between the left and right halves of the display, making a determination of polarity difficult (Fig. 6). Is this device in or out-of-polarity? Our expensive analyzer can't tell us. All three of the polarity testers in the lab agreed that the device was in-polarity, meaning that a positive going signal applied to the + input results in an outward motion of the transducer and an overpressure into the atmosphere. Not quite satisfied with the result, I used Don Keele's tone-burst signal from the Syn-Aud-Con Test CD and an oscilloscope. The assymetrical nature of the signal allows polarity to be easily observed. This test showed that indeed the dome tweeter was reverse-polarity. Don has always cautioned us to "test the testers" and in this case it paid off. Interestingly enough, this is not all that unusual a case, which is why a scope should be a part of every installers test equipment.

In summary, the TEF data was good, but we were looking at device whose phase response made it difficult to determine polarity from the measured phase. I went back and used the scope mode test of the MLS software, which agreed precisely with the oscilloscope. The polarity testers, in this case, got fooled. They are very useful in testing cables and electronics, but should be scrutinized closely for transducer measurements. We consider the oscilloscope and tone-burst the "acid test" and the standard by which to judge the others. In fact, if this were a woofer, a good old 9V battery would have yielded the proper result in the shortest time span.



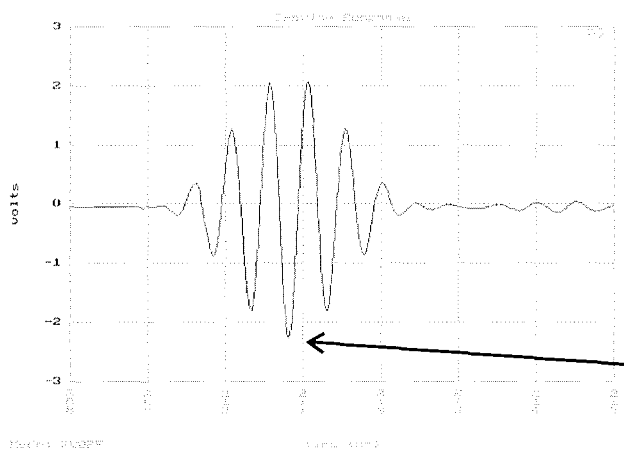
Polarity is difficult to determine for this dome tweeter since the phase does not average around 0 or 180 degrees. A different method is needed for such a device.

Figure 5



The Nyquist display isn't much more help, since the data does not fall clearly into the right half (in-polarity) or the left half (out-of-polarity)

Figure 6

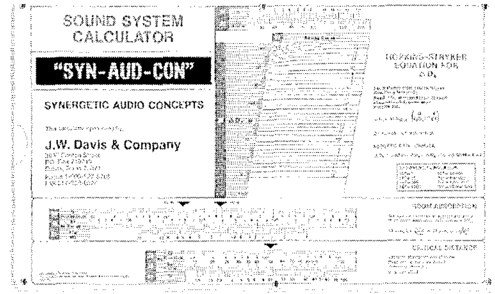


The mid-point of this waveform would be facing upward if the device were in-polarity.

The Final Word

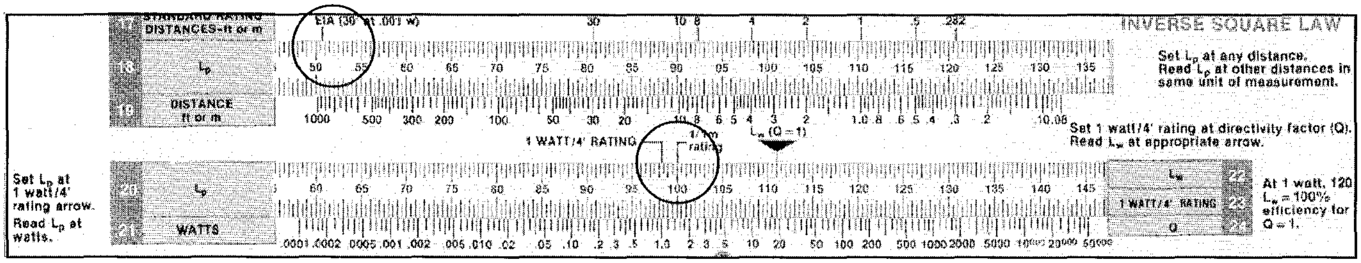
The scope mode of the TEF20 is a good way to capture a one-shot event for further analysis and study. An oscilloscope can also be used to test polarity. In this case, the signal is Don Keele's tone burst test from the Syn-Aud-Con Test CD. It's asymetrical nature makes checking polarity simple and reliable. Note that the dome tweeter is reverse-polarity.

Loudspeaker Sensitivity Ratings

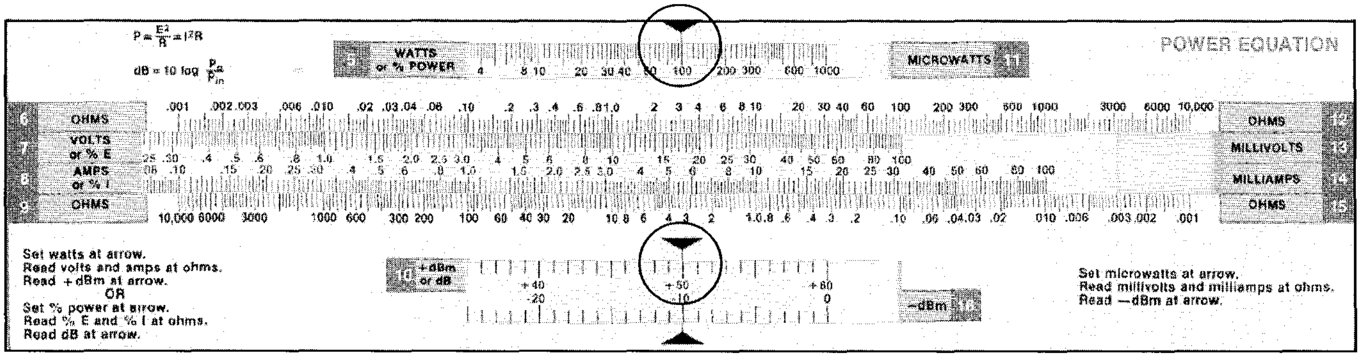


The sensitivity rating of a loudspeaker is useful for calculating the needed amplifier power to produce a given level at a given listener distance. Unless specified otherwise, the sensitivity is measured on-axis and is referred to as the axial sensitivity. These measurements must specify the power used and the distance from the measuring microphone. Most loudspeakers are rated at 1 watt/1 meter. The slide rule is useful for taking this information and using it to determine the needed amplifier power for a project. While there are several ways to do this, we will discuss one that is simple and practical.

1. Consider a loudspeaker with an axial sensitivity of 100 dB 1W/1M. Using the Inverse-Square Law section of the slide rule, place 100 dB on scale 20 opposite the 1/1M marker (about mid-scale). Go to scale 17 and find the marker for EIA (30' at .001 W) and look directly below it to read 50.5 dB. The slide rule has simply converted the measurement distance to 30 feet (which is about 20 dB further than 1 meter using inverse-square law) and the test power to .001 Watt (which is 30 dB less than 1 Watt). The utility of this method is that the sensitivity measurement is now referenced to 0 dBm and 30 feet (or about 10 meters). Any amplifier power in dBm can now be added directly to the EIA rating to yield the level at 30 feet.

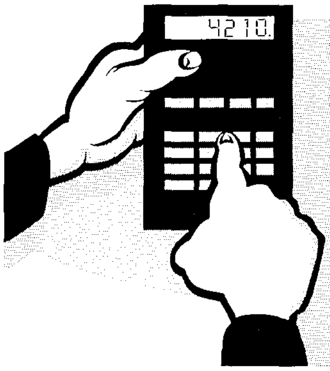


2. To convert amplifier power to dBm, refer to the Power Equation section of the slide rule. Align the amplifier power on scale 5 with the marker (center of scale) and read the level in dBm from scale 10. For example, 100 watts would convert to 50 dBm.



If we were to take this 50 dBm amplifier and connect it to the previous loudspeaker (EIA rating = 50.5 dB) the combination of the two would produce an Lp of 100.5 dB at 30 feet. This reveals an interesting and useful relationship, namely that the 1W/1M sensitivity of a loudspeaker can also be viewed as a 100W/30' rating (within 0.5 dB). This level can now be scaled to any distance and any power. In our example case, we can say that this loudspeaker has a sensitivity of 100 dB at 100W/30'. The level at 60 feet would be 94 dB (down 6 dB due to inverse-square law), and the level at 120 feet would be 88 dB. If this level is sufficient for the design, but 10 dB of headroom is needed, we can simply add 10 dB to the amplifier power, resulting in a 60 dBm (1000 Watt) amplifier.

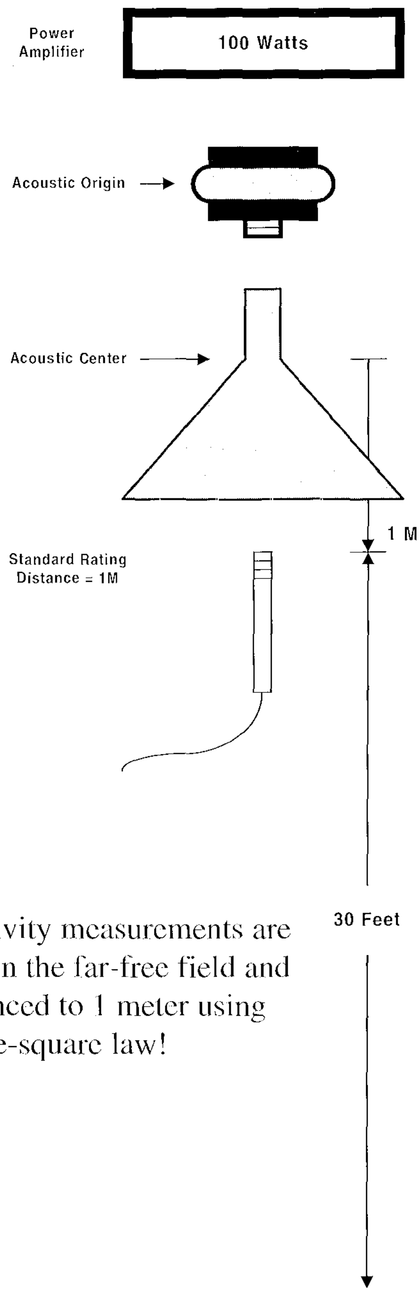
In summary and review, consider the 1W/1M sensitivity to be 100W/30'. This places the rating in the real world concerning listening distance and amplifier power. From this rating, scale the distance and power accordingly for the application at hand. This can be done mentally remembering that 3 dB is twice (or half) the power and that double (or half) of the 30' distance results in a 6 dB change. Don't forget to allow enough headroom and be sure to not exceed the loudspeaker's power rating.



Cranking Thru the Math

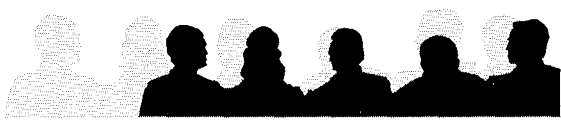
The Method Behind the Numbers for Calulating the Electrical Power Required

Converting Electrical Power to Acoustical Levels



Sensitivity measurements are made in the far-free field and referenced to 1 meter using inverse-square law!

$P_{Amp} = \frac{E^2}{R} = \frac{28^2}{8} = 100\,\text{Watts}$	The continuous average electrical power available from the amplifier.
$P_{Acous} = \frac{\%Eff(P_{Amp})}{100} = \frac{1(100)}{100} = 1\,\text{Acoustic watt}$	The conversion of electrical power to acoustical power
$L_W = 10\log\frac{P_{Acous}}{10^{-12}} = 10\log\frac{1}{10^{-12}} = 120\,\text{dB PWL}$	The expression of electrical power as a level.
$L_W = L_P @ 0.282\text{m}$ $L_P = 120\,\text{dBSPL}$	Coverting L_W to L_P at their common distance.
$\text{Directivity Index DI} = 10\log Q = 10\log(10) = 10\,\text{dB}$	Accounting for the directivity increased offered by a horn (approx. 90x50 coverage).
$L_{P@1M} = L_W + 10\log Q + 20\log\frac{.282}{1} = 120 + 10 + (-11) = 119\,\text{dBSPL}$	Finding the sensitivity at 1W/1M. This value can be scaled to any distance or power.
$L_{P@Listener} = L_{P@1M} + 20\log\frac{3.28}{30} = 119 + (-19) = 100\,\text{dBSPL}$	Inverse-square law level change to the seat of interest.



This procedure is available as a Mathcad template on our World Wide Website at:

http://synaudcon.com/synaud

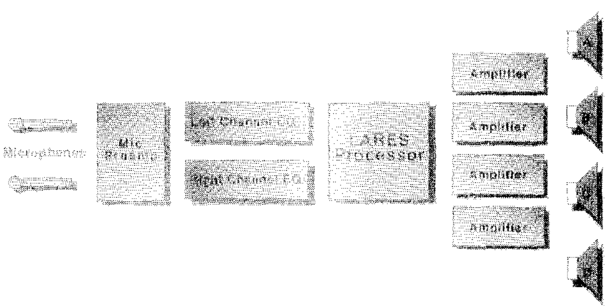
Virtual Acoustics from Wenger

The word "virtual" is quickly becoming a part of the vocabulary of the 90's. Webster says that virtual means

"existing or resulting in essence or effect though not in actual fact, form, or name."

Early in February, I had an opportunity to hear a virtual acoustic environment, in the form of a special room designed and built by Wenger Corporation. Dubbed the "V-Room Studio," this small (physically, at least) space measures 11' x 13'6" x 12' and is designed for acoustic isolation between the occupants and the outside world. For instance, a sound with an L_p of 80 dB on the inside is barely audible at 43 dB on the outside, approximately the sound level in a library. Once inside, the general acoustic feel of the space is that of an anechoic chamber, with the exception of the reflection off of the solid maple floor, which had been intentionally retained to provide acoustic realism for the performer.

The Building Blocks of LARES...
A complete LARES system includes micorphones, one or more LARES mainframes, Equalizers, Amplifiers, and a distributed loudspeaker array.



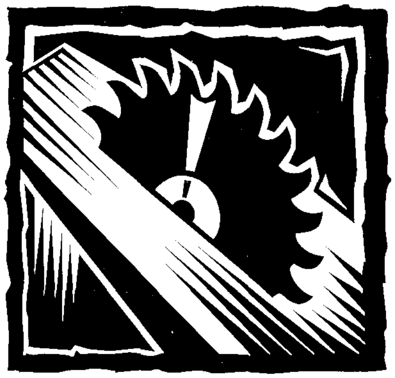
The heart of the V-Room Studio is the LARES system from Lexicon. The LARES (Lexicon Acoustic Reinforcement and Enhancement System) consists of four distributed loudspeaker systems, two equalizers, four channels of amplification, two cardioid microphones, and a mainframe DSP processor with controller. This elaborate system is virtually (there's that word again) invisible to the performer, whose interaction with the system involves only the selection of presets from a wall-mount control pad.

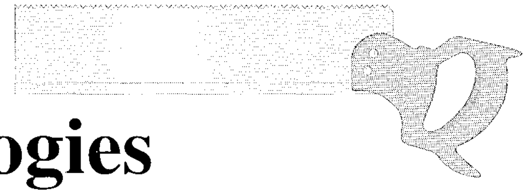
The V-Room Studio is designed to surround musicians with the acoustical response of almost any auditoria so they can improve their overall musicianship. It is a practical laboratory for learning the relationship of space to musical technique.

Steve Barber of Lexicon, and Ron Freiheit of Wenger Corporation provided a demonstration of the facility and an overview of its operation. In addition to some standard presets for different types of spaces, there are also settings for specific rooms that were constructed from impulse response measurements made in those spaces. The utility of being able to practice on the stage of a hall without going there becomes immediately apparent.

The V-Room Studio is an impressive implementation of the LARES system, but applications of this technology are just beginning. The future should allow us to marry the processing power of LARES with the room simulation work of Dr. Ahnert and others to allow effective acoustic auditioning of spaces prior to their construction. Syn-Aud-Con will use a full LARES system in the Design and Advanced Design seminars for the purpose of teaching acoustics. Our attendees will leave with a true understanding of reverberation, early reflections, critical distance, N factor and other parameters of interest to system designers.

Ron Freiheit (Wenger Corporation) and Steve Barber (Lexicon)





Methodologies

During a home remodelling project, I was faced with the dreadful task of installing crown molding around the ceiling in several rooms, including a bathroom with many angles. This is a difficult task for the "occasional" cabinet maker, since two cuts at each joint are required, each related to the other. This makes "cut and try" methods difficult. In an attempt to find a repeatable method of making the cuts, I turned to Mathcad, since the problem was geometric in nature.

Below is the template that I created for determining the correct angles. The math is not my own, but was retrieved from an ancient cabinetmakers resource, a good reminder that virtually all problems in life have already been experienced (and solved) by someone else. The formulas worked, the boards fit, the wife is happy, and life is good.

All of this has served to preface a point that I would like to bring out. As one tours a turn-of-the-century home, crown molding was commonplace, and installed with precision and accuracy that is enviable by even today's standards. Yet, these craftsmen did not have Mathcad, or even powered miter saws. They achieved their results with basic geometric construction techniques that could be executed with a simple straightedge and protractor. Apparently there is more that one way to "execute the mathematics" involved in solving problems, and the mere presence of "high tech tools" does not ensure a superior result. Certainly the greater honor must go to those who can get the same result with the simpler methods.

The design and installation of an audio system is a little like installing crown molding. It can be a difficult task, with many interacting components, yet can be conquered with an understanding of the fundamentals and the intuition that comes from having thought about the problem. Our seminars take a "straightedge and protractor" approach to system design, which we feel provides insights that can be missed with the hi-tech methods alone. Such an approach defines the proper application of the hi-tech tools when they are needed. The main criteria for meaningful system measurements is knowing what to measure, and for sound system/room modeling it is knowing what to model.

Compound Miter Saw Formulas

$A := 4$

$C := 5.62$

$F := 145$

$\alpha := F \cdot \frac{\pi}{180}$

$D := 4$

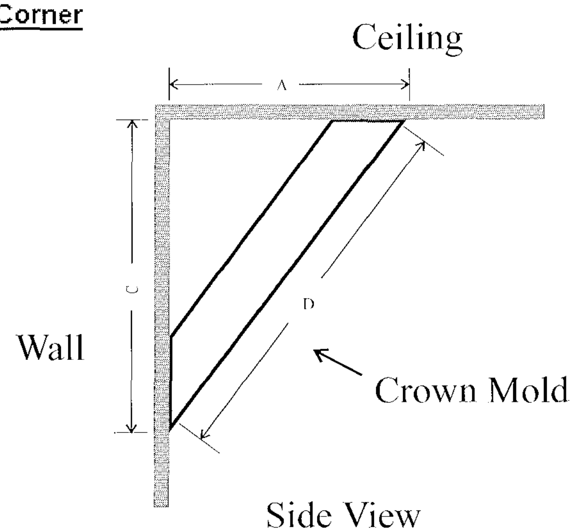
Angle of Inside Corner

$$\operatorname{atan}\left(\frac{A}{C \cdot \tan\left(\frac{\alpha}{2}\right)}\right) \cdot \frac{180}{\pi} = 12.648$$

Miter Angle

$$\operatorname{asin}\left(\frac{D \cdot \cos\left(\frac{\alpha}{2}\right)}{C}\right) \cdot \frac{180}{\pi} = 12.358$$

Bevel Angle



Mark IV Audio Reprints the

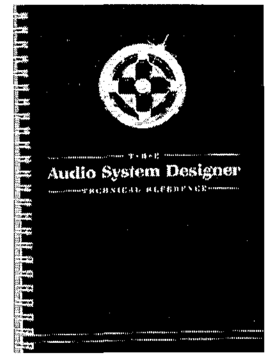
Audio System Designer

Technical Reference

The original release of the Klark Teknik Audio System Designer has become the most referred to "sound system engineering book" that I own. It is been a constant travel companion since I first received it in the '80's.

Audio System Designer has been researched and complied Peter Mapp, a highly respected British acoustical consultant. It brings together information from a wide range of sources that is very useful for the system designer. It includes charts , tables, graphs of all manner of information relating to acoustics and sound system design.

The original edition was published by Klark Teknik and a considerable deal of gratitude must be given to them by all of us in the industry for the foresight they had in preparing this publication. I'm sure I speak for many who would say that this book has been a most welcome edition to their libraries.



Recently a revised edition has been released. In comparing both editions I was surprised to note that none of the information had been updated. On conferring with Peter Mapp he indicated that time constraints meant that the earlier edition was simply reprinted. This is a little unfortunate as there has been considerable advances in some areas that are covered by this book, and it would have been desirable to have this new information included. Notwithstanding the lack of more recent information - all of the information included is still valid, the newer details would have enhanced

Topics covered include:

GENERAL ACOUSTICS
- Typical sound pressure levels
- Typical sound pressure level (SPL) versus sound pressure
- Combining decibels
- Inverse square law
- Wavelength of sound versus frequency
- Musical range versus frequency
- Frequency range of musical instruments and vocals
- ISO preferred octave and 1/3 octave centre frequencies with band limits
- A Weighting
- Background noise design criteria
- NR curves (noise rating)
- NC curves (noise criteria)
- PNC curves (preferred noise criteria)

SOUND INSULATION & ABSORPTION
- Average insulation values (mass law)
- Airborne sound reduction index of solid homogeneous wall (mass law)
- Typical sound insulation performance of building materials compared with mass law
- Resultant sound insulation of composite building structures
- Summary of sound insulation performance for typical building materials - Average 100-3150Hz
- Sound reduction index
- Noise reduction (NR) and transmission loss (TL)
- Room to room transmission via ductwork
- Typical sound reduction data in dB
- Absorption/attenuation of reflected sound components
- Dependence of sound absorption on the angle of incidence
- Panel absorbers
- Helmholtz absorbers
- Porous absorbers
- Bass traps
- Average absorption coefficients

ROOM ACOUSTICS
- Room modes (eigentones)
- Optimum room ratios
- Sound pressure and power level
- Relation between reverberation time, volume and absorption
- Room constant versus surface area and absorption
- Critical distance as a function of room constant and directivity
- Reverberation time formulae
- Reverberation time criteria
- Reverberation time data for concert halls
- Typical reverberation time design targets

PHSYCO-ACOUSTICS AND SPEECH INTELLIGIBILITY
- Dynamic range of hearing
- Equal loudness contours
- Typical hearing loss with age (presbyacusis)
- Haas effect
- Directional characteristics of human voice
- Male speech spectra
- Female speech spectra speech level
- Speech interference levels
- Speech privacy (approximate guidelines)
- Percentage loss of consonants (% Alcons PB wordlist)

SOUND SYSTEM ENGINEERING
- Ohms law

that which is included.

Sadly the new edition is not presented in the same format as the original. The original used an ISO A5 size (US 1/2 letter size) page in a 3 ring binder, which allowed plenty of space for inclusion of additional information. The new edition, whilst the same page size, is spiral bound and therefore does not enable adding to the information included.

The good thing about the size is that it now fits nicely into a briefcase, which means that you will be much more likely to have the information available when you need it.

These *minor* reservations about the reissue of the audio system designer in no way counteract its value as a source of useful information for all designers. I highly recommend this book to all who do not have a copy.

We now eagerly await Peter Mapp's new publication "Handbook of Jokes and Amusing Stories for the Travelling Acoustical Consultant"

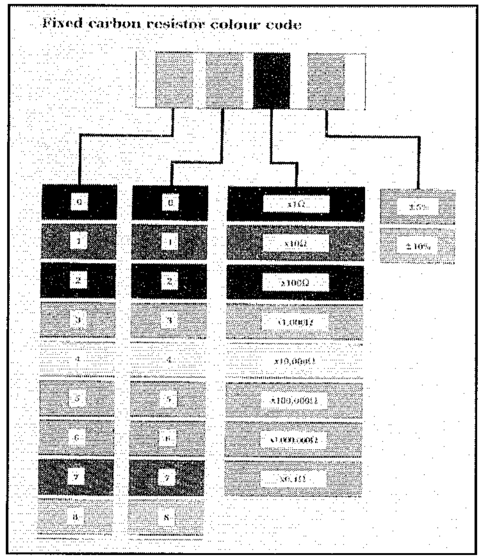
Reviewed by:

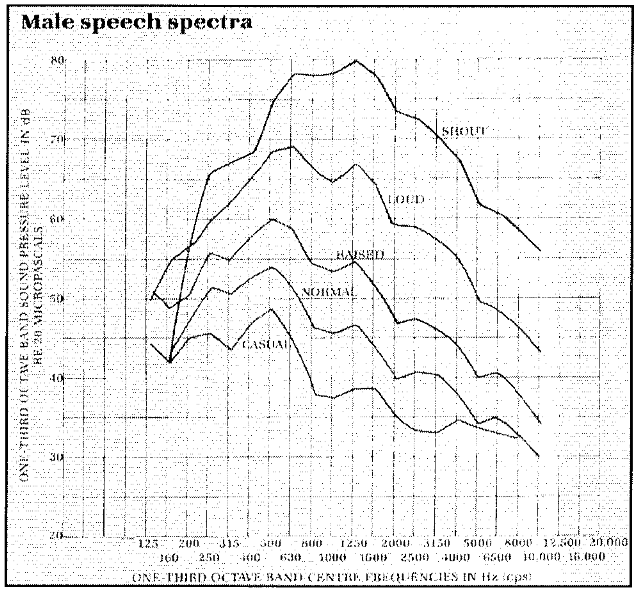
Larry Elliott - Auckland, New Zealand

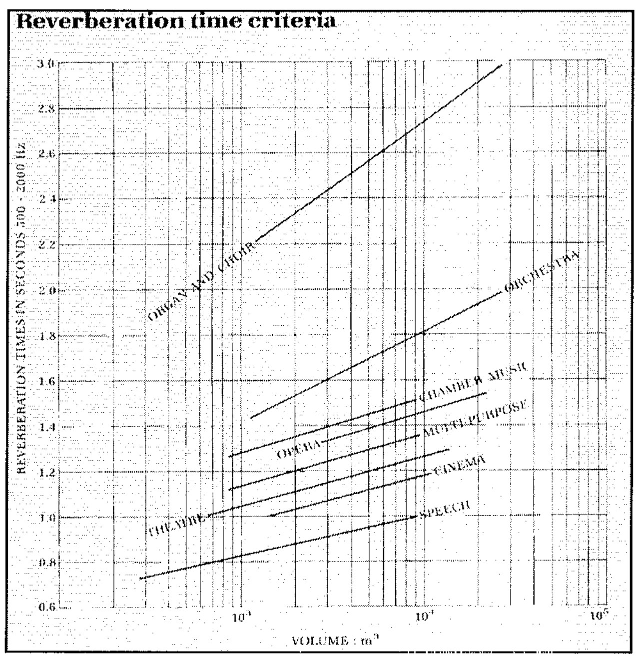
The Audio System Designer is available from:
Synergetic Audio Concepts
Mark IV Audio

Basic electronic pad circuits
Fixed carbon resistor colour code
Electronic noise measurement curves
Distortion (THD) conversions
Voltage versus dBv and dBu/dBm (600W)
dB ratios
Voltage and power conversions
Acoustic power from a loudspeaker
Power versus SPL
100 volt line loudspeaker systems
Loudspeaker line losses
Distance, velocity and time
Wind and temperature gradients
Atmospheric air attenuation
Sound propagation through trees
Speed of sound in various media (at 21 degrees Celsius)
Effects of time delay
Percentage disturbance
Combfilter peak and nulls
Speech integration and intelligibility
Ceiling loudspeaker coverage
Sound system equalisation curves
Articulation index
Relationship between articulation index, communication and speech privacy
Dependence of the percentage syllable articulation on

PHYSICAL DATA
Metric conversion tables
Conversion Factors
Audio connectors







Crossover Align/ment Using a Real-Time Analyzer

Last quarter we did a study of improving a two-way system using signal delay. The measurements were performed on a TEF analyzer, and many have asked if the same thing can be done on an RTA. This study should help answer that question.

Multi-way systems present some special problems to the system designer. A major one is achieving a smooth transition between devices covering different pass bands, such as matching a 15" loudspeaker to a midrange horn. The problem lies in the fact that the two devices are of different physical size, and as a result emit sound from different points in space. Depending upon the crossover frequency, the transition area can contain a sizable notch in frequency response. One solution in the past has been to reverse the polarity of the horn. This can smooth out the crossover region, but can arguably cause other problems (transient response, harmonics reverse-polarity with fundamentals, etc.). The best solution is to use a precision signal delay to align the devices at the crossover frequency.

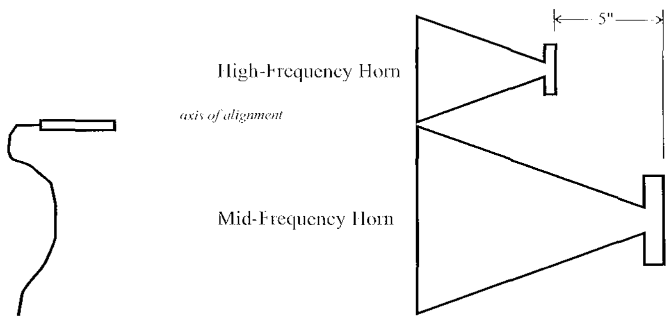
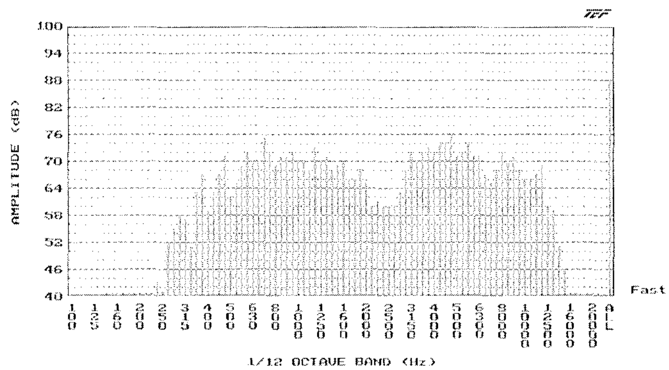


Figure 1 shows the response of a two-way loudspeaker with mid-frequency and high-frequency horns. The mid horn is about six-inches deeper than the high-frequency horn. Note the notch at crossover caused by the differing acoustic origins of the two devices (Fig. 3). Intuition tells us that in this case the mid horn is probably arriving before the horn at the crossover frequency (3kHz). As such, some delay for the high-frequency horn becomes necessary.

To correct this problem, a precision signal delay has been placed in-line with the high-frequency horn. The output of the system is being viewed midpoint between the two horns on-axis on a real-time analyzer with 1/3 octave resolution. The delay is increased until the notch smoothes out. A quick-check of the delay setting reveals that it is about 5 inches or about 0.5 ms. Higher resolutions (1/6 octave and 1/12 octave) are readily



RTA's with 1/12 octave resolution are now available

available on some analyzers, but to date 1/3 octave units dominate the marketplace.

A couple of things to keep in mind...

First, most digital delays have several milliseconds of delay inherent to them, the cost of processing the signal. The best choice is to use a programmable crossover with precision delay, which will reference all of the outputs to the same point in time.

Secondly, this technique has not eliminated the problem at crossover. We have simply moved the axis of misalignment away from the on-axis position. If the mic were moved up or down at small increments, we would find that the notch reappears. This technique assures that the energy is going where the loudspeaker is aimed, and hopefully the notch is now oriented toward an area not adversely affected by it.

Improvements over a wider area can be realized by coaxially mounting the devices, with the high-frequency horn actually placed inside of the mid-frequency device (assuming that the physical construction of both of these devices makes this possible).

If the only tool at hand is a sound level meter, proper synchronization can be accomplished by using band-limited pink noise centered at the crossover frequency and adjusting the delay for maximum reading on the meter. It is also possible to do it by ear (a trained ear!) by aiming the loudspeaker at a hard wall, standing behind the loudspeaker and listening while adjusting the delay. Use pink noise as the stimulus. The proper delay setting will announce itself by the strong reflection coming off of the wall.

Today's modern processors have given us many tools for aligning multi-way systems at crossover. The goal for any method employed is always to make the system response appropriate for the application. Sound system design has been described as the "fine art of compromise". In this case, we prefer a correct response on-axis (at a listener seat) and elected to align at that point, placing the axis of misalignment in a hopefully unnoccupied area, or at least one that is closer to the stage.

Seeing the Difference...

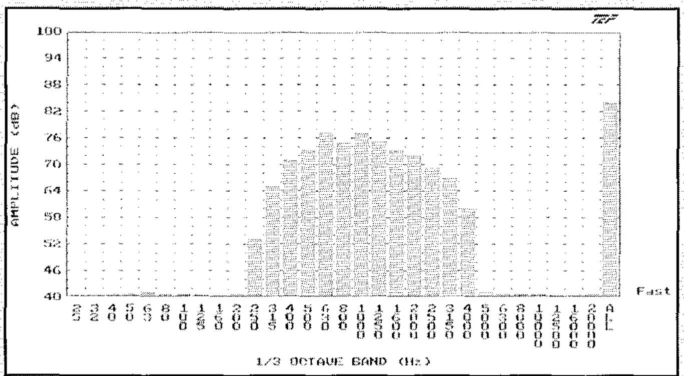


Figure 1 - The response of a mid-range horn as viewed on a 1/3 octave real-time analyzer. This would be an excellent device for a voice-only application, such as a paging system, but it lacks the high-frequency response needed for music reproduction.

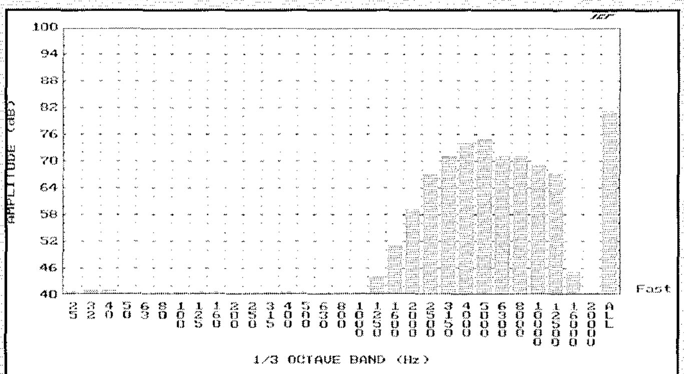


Figure 2 - The response of a high-frequency horn as viewed on a 1/3 octave real-time analyzer. This device should be useful in providing the needed "sibilance" to complement the device in figure one. The problem now is in getting these two devices to sum properly at the crossover frequency (3 kHz).

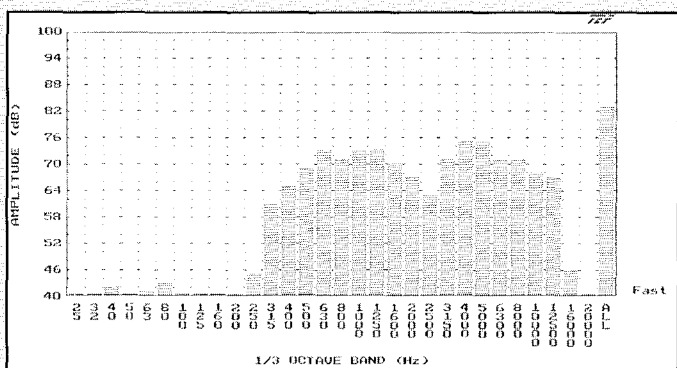


Figure 3 - The combined response of the mid and high-frequency horns as viewed on a 1/3 octave real-time analyzer. Note the large dip in the crossover region caused by phase cancellation. This occurs because the sound is not arriving coherently from each device at the point of observation.

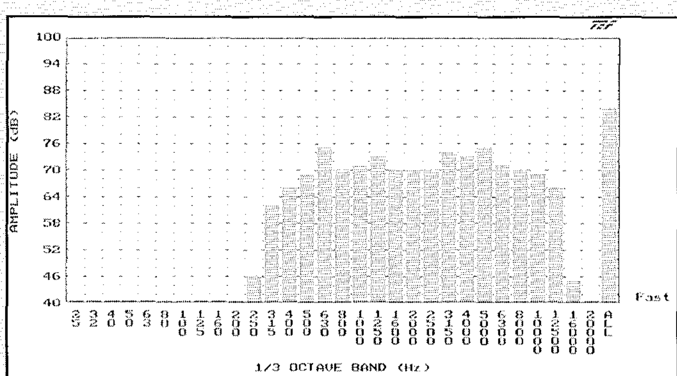


Figure 4 - A precision signal delay (20 us/step) is placed in-line with the high-frequency horn, since its sound is arriving first. A delay of about .5 ms (5 inches) was adequate to improve the response at crossover. The improvement was quite audible on pink noise and program material.

Anatomy of the Decibel

Part II - More on References

Last quarter we examined the mathematical process involved in expressing a relationship between two quantities in decibels. Without the dB, the numbers that we would have to work with in describing audio systems would be huge and cumbersome. The dB was developed to allow these large numbers to be "compressed" down to smaller ones, and involves three operations to achieve the conversion. First and foremost, the numbers being expressed in dB must be a "power like" quantity. This means that for an electrical signal we can express power, resistance, or the squared voltage or squared current in dB. This can be remembered by referring to Ohm's Law, which describes the relationship between voltage, current, and resistance in regards to power.

$$W = \frac{E^2}{R} = I^2R$$

where W is power
E is volts
I is current
R is resistance

The dB can be used properly to describe the relationship between any two powers, voltages, currents, or resistances. These quantities are normally measured at the same point in the circuit. For example, to determine the difference in level at the listener for two different power amplifiers, we can write the following:

$$dB = 10\log\frac{W_1}{W_2}$$

Additionally, if we wished to describe the level change at the listener for an decrease in output voltage from the mixer, we can write:

$$dB = 10\log\frac{E_2{}^2}{E_1{}^2} = 20\log\frac{E^2}{E^1}$$

In this case, the ratio of the two voltages describe the level change at the listener, since they were acquired at the same point in the circuit.

It is often convenient to compare measured powers voltages with a standard reference rather than with each other. For power, the most commonly used reference is 1 milliwatt, with the outcome expressed in dBm (dB ref. to 1 mW). Expressing amplifier powers in this manner will yield the difference in level at the listener caused by increasing or decreasing the amount of watts. For example, going from 100 watts to 250 watts will cause the following level change at the listener:

$$dB = 10\log\frac{250}{100} = 4\,dB$$

$$\text{Level One} = 10\log\frac{250}{.001} = 54\,dBm$$

$$\text{Level Two} = 10\log\frac{100}{.001} = 50\,dBm$$

$$\text{Level One} - \text{Level 2} = 54 - 50 = 4\,dB$$

Once the amplifier level is expressed in dBm, it can be used to modify the sensitivity rating of a loudspeaker (see Slide Rule Quarterly, this issue) and determine the electrical power required for a sound reinforcement system.

Some care must be taken when applying term "level" to voltages. Levels, by definition, are always power ratios (watts) with one of the powers being a standard reference power, such as 1 milliwatt for dBm. Voltages (and pressures) are sometimes expressed as levels, but this only appropriate when the impedance is common to both voltage or pressure quantities, such as with pressure measurements made with a sound level meter, or in contant voltage signal processing chains. In such cases, it is the change voltage amplitude that determines the level change at the listener.

The IEEE standard reference for voltage is the dBV, which is dB referenced to 1 volt. The NAB standard is the dBu, which is dB referenced to 0.775 volts. Either of these references can be used to calculate the level change at the listener caused by a change in voltage at some point in the circuit. For instance,

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suppose the output voltage of a mixer were raised from 2 volts to 5 volts. Assuming sufficient amplifier power is available, the level change at the listener would be:

$$dB = 20\log\frac{5}{2} = 8\,dB$$

$$\text{Level One} = 20\log\frac{5}{1} = 14\,dBV$$

$$\text{Level Two} = 20\log\frac{2}{1} = 6\,dBV$$

$$\text{Level One} - \text{Level Two} = 14 - 6 = 8\,dB$$

or, using the dBu:

$$\text{Level One} = 20\log\frac{5}{.775} = 16.2\,dBu$$

$$\text{Level Two} = 20\log\frac{2}{.775} = 8.2\,dBu$$

$$\text{Level One} - \text{Level Two} = 16.2 - 8.2 = 8\,dB$$

Notice that the same answer was arrived at with three different methods. If one understands processes behind the methods, the actual one chosen becomes a matter of convenience or convention.

If a traveller understands several languages, he or she would use the one appropriate for where they happen to be. Our classes teach the principles involved with each, so that our attendees become conversant with the dBm, dBV or dBu.

Those who complete our Level Two class leave with the ability to work from the talker to the listener using the decibel. This marvelous tool allows most any system design problem to be simplified and solved with simple addition or subtraction.

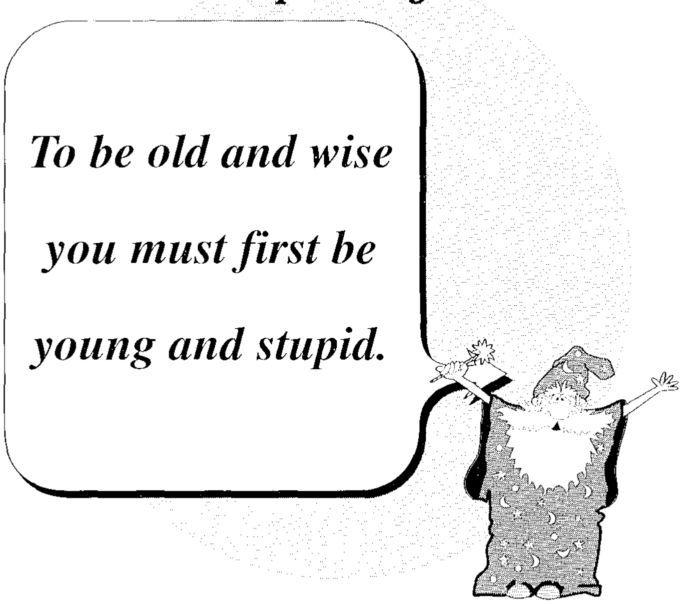
When one considers the various quantities encountered when working from the input to the output of a sound system design; pressures, powers, voltages, impedances, etc., the decibel becomes a much needed "common denominator". It allows us to know what happens at the output of the system based upon something that happens earlier in the signal chain. The effect of changing talkers, mics, mixers, amps, etc. can be quickly and accurately predicted. For acoustics, it is an ideal tools for establishing the level and direct-to-reverberant ratios at listener positions, and how these are affected by changes in transducers and/or signal processing devices. It is a valuable tool for assessing and predicting the speech intelligibility of a system/room combination.

Keep in mind that the decibel, like human language, was developed to make things easier, not more complex. That is why being "dB minded" is of prime importance for system designers and operators.

Quantity	Reference	Label
Sound Power	10^{-12} Watt	L_W
Sound Intensity	10^{-12} W/m^2	L_I
Sound Pressure	.00002 Pascal	L_P
Electrical Power	.001 Watt	dBm
Electrical Voltage	1 Volt or .775 Volts	dBV or dBu

Standard references for using the decibel in audio system work.

The Old Philosopher Says...



Best T-Shirt Award

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.

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

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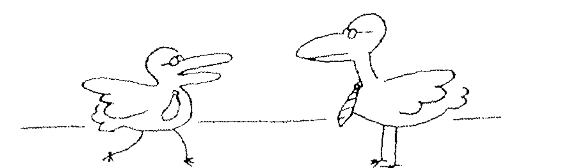

Bad news, chief. The early birds got the worms!

Quote of the Day...

"If I were reincarnated, I would want to come back as a buzzard. Nothing hates him or envies him or wants him or needs him. He is never bothered or in danger, and he can eat anything."

William Faulkner

Here are a couple of "buzz words" that I keep running across in industry articles...

qual·i·ta·tive (kwŏl′ĭ-tā′tĭv) *adjective*
Abbr. **qual.**
Of, relating to, or concerning quality.
[Middle English, producing a primary quality, from Medieval Latin *quālitātīvus*, from Late Latin, qualitative, from Latin *quālitās*, *qualitāt-*, quality. See QUALITY.]
— **qual′i·ta′tive·ly** *adverb*

quan·ti·ta·tive (kwŏn′tĭ-tā′tĭv) *adjective*
Abbr. **quant.**
a. Expressed or expressible as a quantity. **b.** Of, relating to, or susceptible of measurement. **c.** Of or relating to number or quantity.
[Medieval Latin *quantitātīvus*, from Latin *quantitās*, *quantitāt-*, quantity, from *quantus*, how great. See QUANTITY.]
— **quan′ti·ta′tive·ly** *adverb*
— **quan′ti·ta′tive·ness** *noun*

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