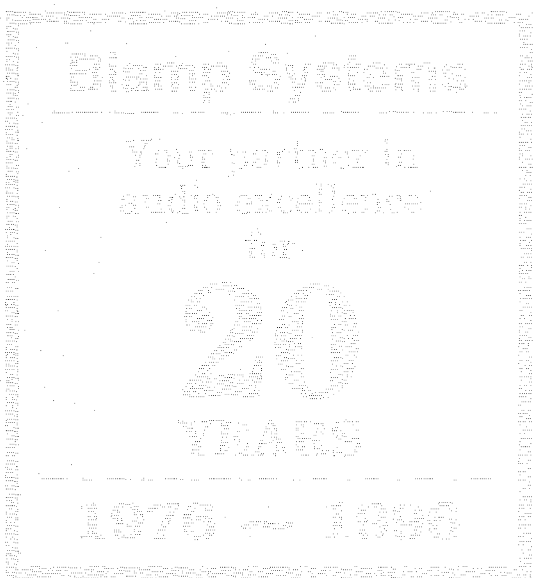
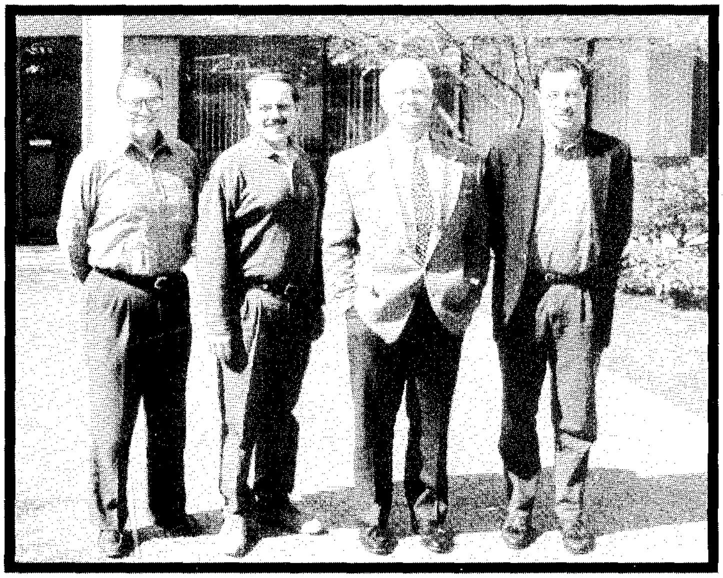
SYNERGETIC SYN AUD CON AUDIO CONCEPTS

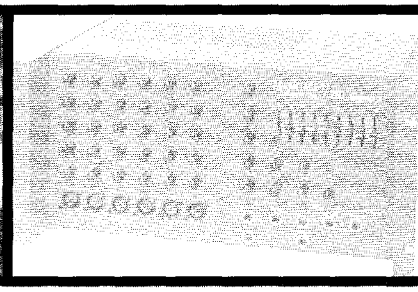
newsletter

Volume 24, Number 3
Summer 1996

Biamp Systems Celebrates 20th Anniversary







biamp AMERICAN SOUND CRAFTSMANSHIP

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, Brenda Brown
Design and Layout: Pat Brown, Brenda Brown

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Tech Topic "A Method of Processing Real Data to Determine Energy Envelopes" by Dr. Sidney Bertram

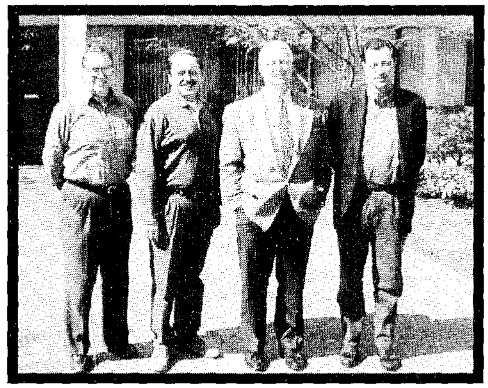
When do I renew? - You can check to see when your subscription will expire by checking the mailing label on the envelope in which your newsletter was mailed. In the upper righthand corner beside the name, a date will appear (i.e., 7-94). This means you will receive your last issue with that quarter's 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.

POSTMASTER: Send address changes to Synergetic Audio Concepts, 8780 Rufing Rd. Greenville, IN 47124

Biamp Systems Celebrates 20th Year

Biamp was founded the year that I graduated from high school, and I can remember seeing their exhibit at the NAMM show that year. Their products were innovative and of higher quality than most other offerings in the musical instrument marketplace. I jumped at the chance to become a Biamp dealer when I went into business a few years later. Biamp was always at the top of my list when it came to customer support, as I am certain that I bugged their tech support people to no end. Here, 20 years later, it is good to have the relationship with Biamp continue through Syn-Aud-Con.

Pat Brown



Pictured from left to right are:
Ralph Tennant (VP Manufacturing),
Jerry Payette (VP Finance),
Ralph Lockhart (President),
Ron Camden (VP Sales/Marketing)

BIAMP SYSTEMS was founded in 1976 to provide high-performance sound reinforcement equipment to professional and semiprofessional musicians. BIAMP SYSTEMS soon earned a worldwide reputation for engineering and manufacturing innovative and reliable professional audio equipment with the finest sonic performance in its class. In the two decades since, the company's product line expanded to include equipment for recording and playback of musical material, as well as for live sound reinforcement.

In recent years the company has applied its skills to the introduction of ADVANTAGE products, a growing line of quality audio equipment specifically designed for installation by professional sound contractors. The same high levels of innovation, reliability and sonic excellence have earned the ADVANTAGE products a wide acceptance among the most demanding sound contractors and audio consultants.

BIAMP SYSTEMS manufactures automatic and manual rack mount mixers, power amplifiers, rack and wall mount powered mixers, programmable and graphic equalizers, digital delays, devices for signal routing and distributing, gain management and remote control for live sound reinforcement and entertainment playback systems.

Today, BIAMP SYSTEMS is forging the future with cost-effective digital audio, and digitally controlled audio products, and remotely controlled products, including the capability of infrared remote control. User-oriented convenience and simple setup programming provide inherent reliability that is easy to install. These innovative technologies offer the audio professional unique and flexible approaches to product application and provide the novice user simplified system operation.

Located in Portland, Oregon, the company staff consists of approximately 65 engineers, technicians, skilled metalworking and electronic assembly craftspersons, and sales and applications professionals. BIAMP SYSTEMS is the professional audio products affiliate of Rauland-Borg Corporation, the technology leader in the field of intercommunication systems for schools, hospitals, prisons, business and industry.

The company's products are sold by some 600 pro-audio and sound contractor dealers in the United States and by distributors and dealers in over 35 countries around the world. Installed systems include houses of worship, sports arenas, legislative chambers, courtrooms, airports, schools, factories, offices, restaurants and clubs, museums, auditoriums and theaters all over the world

biamp AMERICAN SOUND CRAFTSMANSHIP

Carolyn Davis to Receive AES Fellowship

Carolyn Davis will receive an AES Fellowship on November 8, 1996 at the Los Angeles Audio Engineering Society Convention

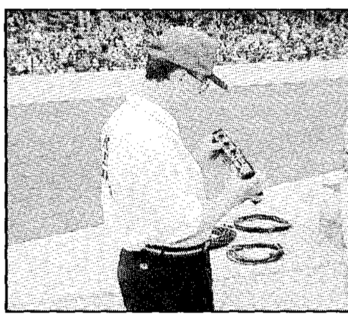
Thousands of Syn-Aud-Con grads saw the "mothering" side of Carolyn during our 23 years of Syn-Aud-Con seminars. The role not visible to the classes was her meticulous attention to technical details, the unswerving integrity of inquiry into often contentious subjects, and the insatiable desire to share hard-won knowledge with others. We like to think that the AES Fellowship award to Carolyn is the recognition for these qualities.

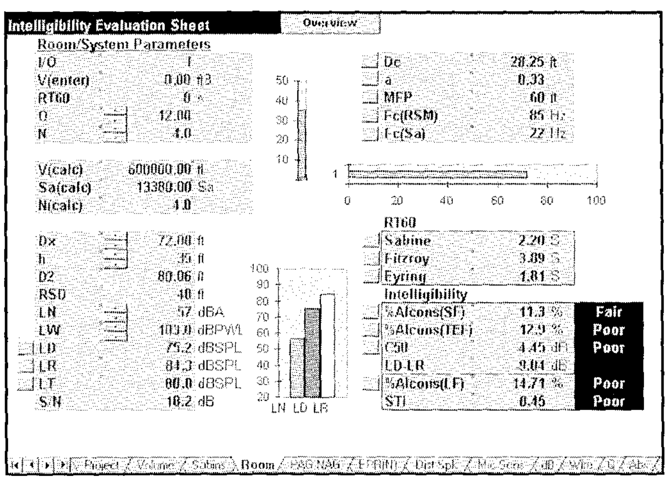
Don Davis

Some Fellow Hoosiers...

We were pleased to have our Indiana/Kentucky representatives, Kurt and Maria Gish of A/V Marketing, attend the Sound System Operation class in Southern Indiana.

Headquartered in Indianapolis, A/V Marketing is a company that we hear lots of good things about. They now have a new facility with more room (and probably more headaches!). Kurt is also on the Sound Staff at the Indianapolis Motor Speedway, and is pictured aiding in the hookup of the microphone used to officially start the race (see article on page 13).





New Spreadsheet for Level Two Class

The Salt Lake City class will mark the first use of our new sound system design spreadsheet, which will allow our attendees to implement what they have learned in the class with simplicity and accuracy. The spreadsheet runs under Microsoft Excel for Windows 95, and contains many of the nested routines used in our presentations, something that we have had numerous requests for. The spreadsheet will be beta tested by our class attendees through the remainder of the year.

THE CALF PATH

One day, through the primeval wood, a calf walked home as good calves should;
But made a trail all bent askew, a crooked trail as all calves do,
Since then three hundred years have fled, and I infer the calf is dead.
But still he left behind his trail, and thereby hangs my moral tale.

The trail was taken up next day by a lone dog that passed that way;
And then a wise wether sheep pursued the trail o'er vale and steep,
And drew the flock behind him, too, as all good bellwethers always do,
And from that day, o'er hill and glade, through these old woods a path
was made.

And many men wound in and out and dodged and turned and bent about,
And uttered words of righteous wrath, because 'twas such a crooked path,
But still they followed, (do not laugh), the first migrations of that calf,
This forest path became a lane, that bent and turned and turned again.

This crooked lane became a road, where many a poor horse with his load,
Toiled on, beneath the burning sun and travelled some three miles in one.
And thus a century and a half, they trod the footsteps of that calf.
The years passed on in swiftness fleet; the road became a village street.

And this, before men were aware, a city's crowded thoroughfare.
And soon the central street was this, of a renowned metropolis.
And men two centuries and a half, trod in the footsteps of that calf.
A hundred thousand men were led, by one calf near three centuries dead.

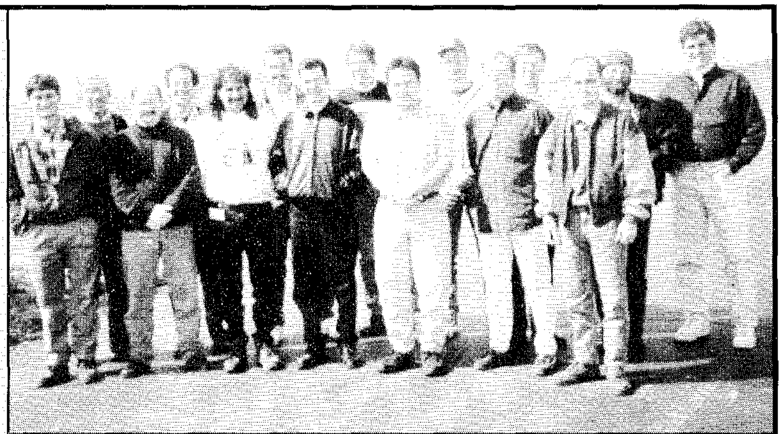
For men are prone to go it blind along the calf-paths of the mind,
And work away from sun to sun to do what other men have done.
They follow in the beaten track, and out and in, and forth and back,
And still their devious course pursue, to keep the path that others do.

They keep the path, a sacred groove, along which all their lives they move,
But how the wise old wood-gods laugh, who saw the first primeval calf.

from RON STEINBERG
(the simple teller of another's tale)

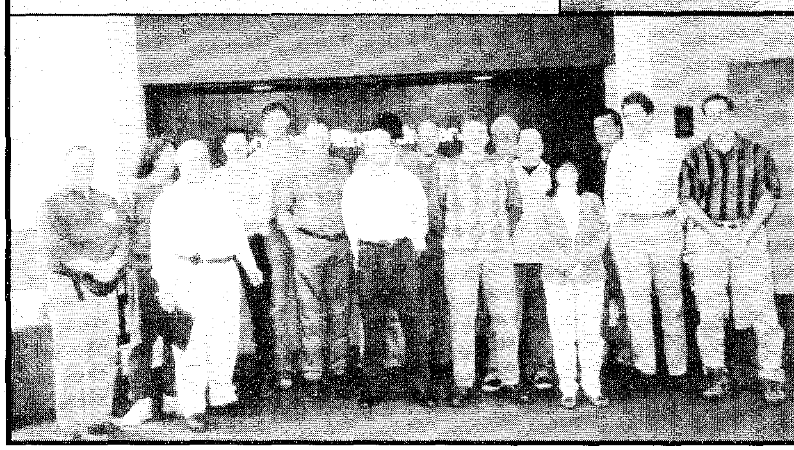
Thanks for your support...

Sound System
Design Seminar
Southern Indiana



Sound System
Operation Seminar
Southern Indiana

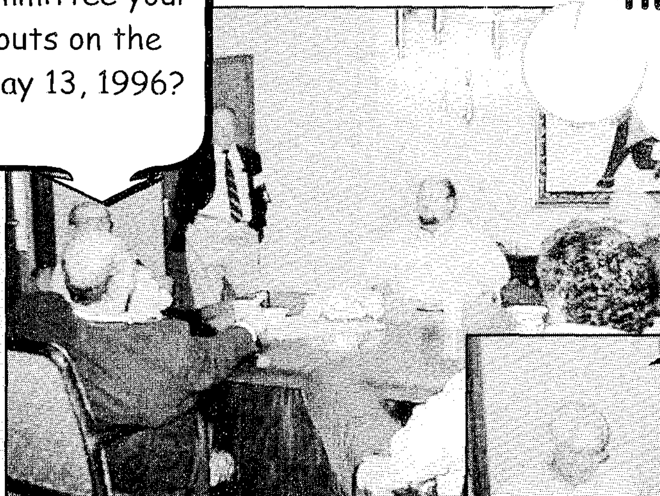
Sound System
Operation Seminar
Portsmouth, Virginia

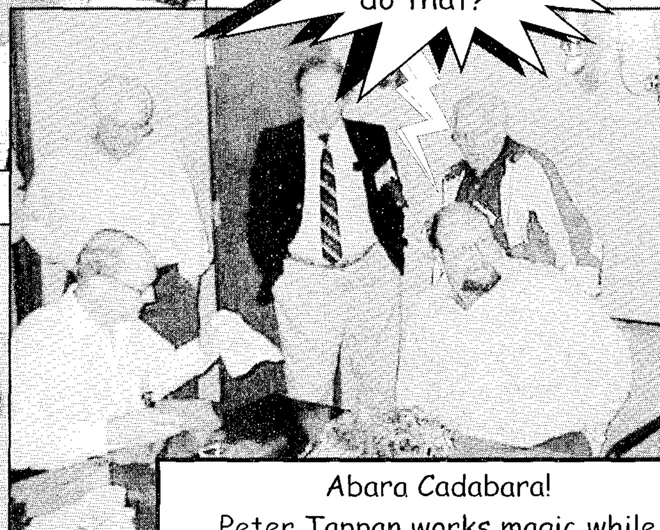


Sound System
Design Seminar
Chicago, Illinois

Time for another episode of

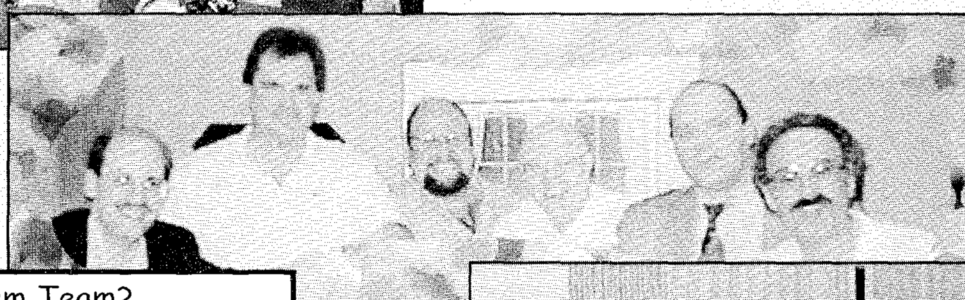
The JW Davis/Syn-Aud-Con Hospitality Suite



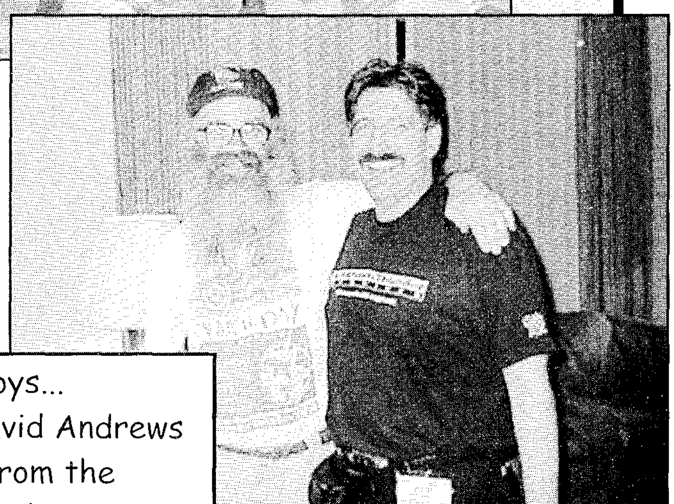


Abara Cadabara!
Peter Tappan works magic while Jack Tucker, Neil Muncy, Glen Ballou and Sam Berkow observe.

Dieter Michel from Pro Sound Magazine, Germany.

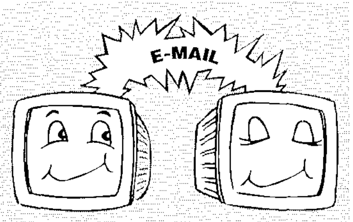


DSP Dream Team?
While we can't reveal their names, their initials are Larry Shank, Jim Bumgardner, Sam Berkow, Don Eger, Blair McNair, and Farrel Becker.



Audio Poster Boys...
You might recognize David Andrews and John Murray from the Vega and TOA ads.

Synergy on the Internet



Our e-mail listserv has proved to be a valuable way to get information about specific subjects. Here is a recent thread that you may find interesting.

Question:

Does anyone know of a device that will digitally remove background noise (tape hiss) from an old recording. Or possibly someone who provides this service. I have been approached with this question and cannot think of anything outside of Dolby NR or an expander. -Mike

Replies:

SAE used to make a "pop" and "click" eliminator which actually worked quite well. When it sensed a pop it substituted a bit of delayed material in its place!

Unfortunately, I can't help in finding one. Perhaps this will jog someone else's memory.

Frank Ostrander
71401.1153@compuserve.com

Know anyone in the FBI?....
Todd Welti
welti@welti.seanet.com

Getting On the List...

If you have a computer, modem, email address, and current newsletter subscription, you can be a part of the Syn-Aud-Con listserv. The listserv consists of several hundred audio professionals worldwide who can drawn upon each others knowledge and experience with this valuable tool. To send a message to the group, just create an e-mail message and send it to:

synaudcon@iglou.com

The entire list will receive the message, and those who can help are able to respond by returning a message. Responses can be to the entire group (encouraged) or specifically to the individual who asked the question.

To subscribe to the listserv, notify me by e-mail (patb@synaudcon.com) and I will sign you up (It is a moderated list).

Dear Michael,

Took me a while to put my hands on the info, anyway, check out a Philips product called the 'Sound Enhancer PR5021'. I saw a prototype at the AES Expo in NY (3 years ago?), and I seem to have read somewhere that they have started shipping a domestic, as well as a professional, version of the unit (the difference being connection standards). Marketed as a device to bring vinyl up to the quality of a CD recording it probably falls short (when will Philips learn?) However, it does have scratch and noise removal. How good is it? Not as good as more expensive solutions and better than cheaper solutions. Now isn't that novel!

The further info box on the flyer lists Philips Key Modules, DCC Mastering and Duplication, Building SAN 4, PO Box 218, 5600 MD Eindhoven, The Netherlands.

Tel +31 40 73 34 55. Fax +31 40 73 67 14

Other Ideas you have probably received already such as Sonics 'No Noise' and Cedar units.

Best of luck - let Syn-Aud-Con know the results of your search.

David E. Smith
David E. Smith & Associates
70232.2772@compuserve.com

Michael,

Two (non-digital) tricks I have used to some degree of success.

A) use an equalizer (stereo) to roll off all frequencies above 4 kHz or so. Take the output from the EQ and run it into an Aphex Aural Exciter. Use the Aphex to artificially "rebuild" the upper harmonics EQ'ed out, minus the noise (if hiss is what you are mostly concerned with).

B) Behringer Electronics (a division of Samson) makes a single-end noise reduction unit that uses a combination of a noise gate and a dynamically controlled low pass filter that, with some experimentation, does a real nice job of removing (hiding, masking, reducing) upper frequency hash.

My favorite?? The Aphex. I have used the Aphex on some really severely noisy tapes. I had to roll off everything above 1 kHz on a particularly bad, old, dropout laden, jazz archival recording. The results --- were phenomenal, considering the decay of the original recording. The Behringer does a REAL nice job on normal to moderately noisy tapes. Hope this helps.

Gary A. Schmitt
Revelation Sound, Inc.
Miami, Florida.
74041.1651@compuserve.com

Thanks to all who responded to Mike's request.

Pat Brown

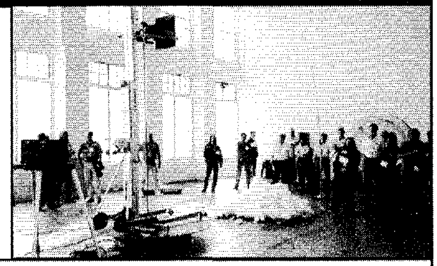
Sound System Training Seminars

Please see the back page for current schedule...

Sound System Operation

Syn-Aud-Con Level One

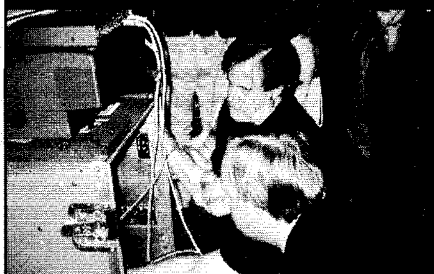
This two-day entry level course is design to familiarize the participant with the basic concepts of sound reinforcements systems. The seminar focuses on the basic workings of a sound system, and how to bring the existing sound system to its maximum potential. If your sound system duties primarily include 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



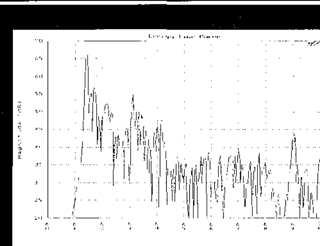
For 23 years, Syn-Aud-Con has been providing training in designing sound reinforcement systems. The Design seminar is the most comprehensive short-course in system design training available today. The contractor, consultant, and advanced system operator will enjoy this three-day audio and acoustic experience, 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

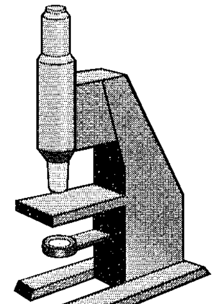
Audio and acoustic technology is constantly evolving, and the Advanced System Design Seminar is designed to keep the professional or serious amateur current with the latest standards and techniques. This three-day seminar places a strong emphasis on the role of the computer as a design, measurement and 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...

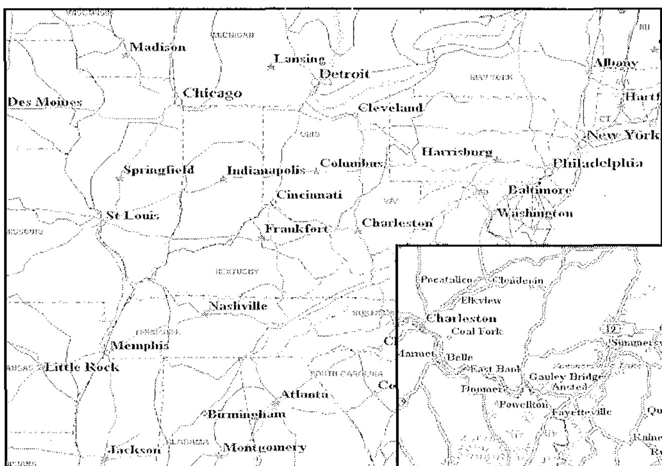
The Issue of

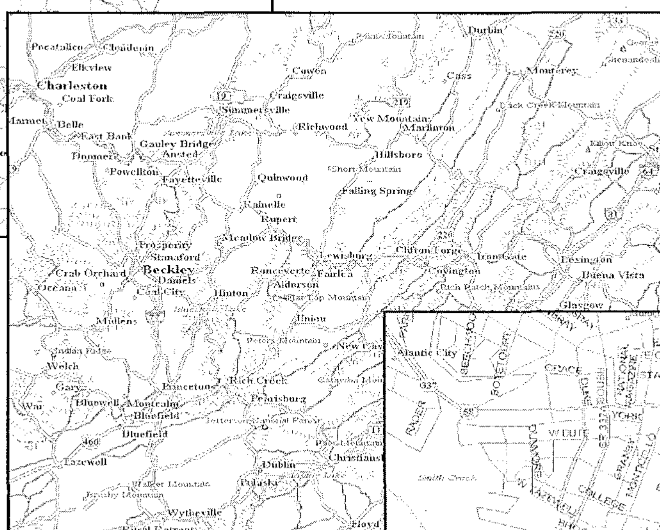
Resolution

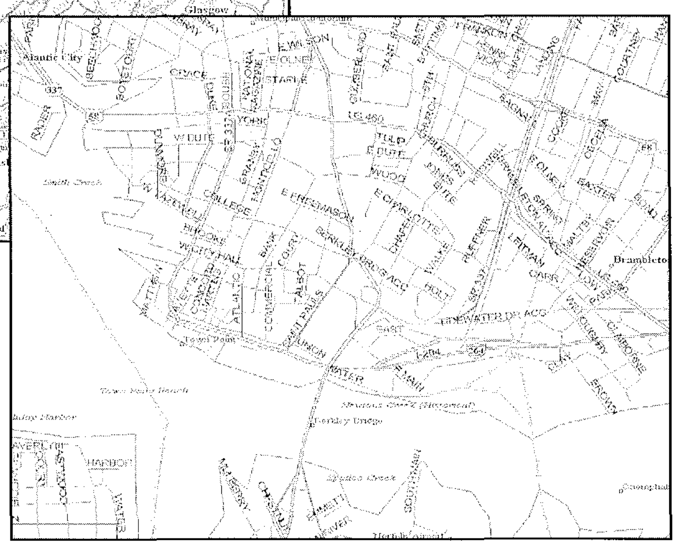


The issue of resolution comes up quite often in sound system design and measurement. It is an important issue in that it can dramatically effect the way that we implement and calibrate systems. This article will examine the issue of resolution from several perspectives, including some non-audio ones. Hopefully the insight gained will improve our ability to implement quality audio systems.

Resolution means the ability to "resolve" or distinguish detail. As is true of many areas of life, more is not necessarily







better. What we are really after is an "appropriate" resolution for the task at hand. Consider a road map. If I am trying to find the best interstate highway to get from Indiana to Norfolk, Virginia, a great deal of resolution is not needed, and a map of the entire USA would suffice. En route to Virginia, if I were trying to find the next exit on the highway, a map with greater resolution would be needed. And finally, if I were trying to find my way to the Holiday Inn in downtown Portsmouth, an even more detailed map would be required. In each case, an appropriate resolution is called for to get the job done.

Note that in the maps pictured, there is a trade-off between resolution and the total area that can be viewed. In a 2" x 3" square, it is simply not possible to display the detail needed to see the whole country and the detail needed to find a Holiday Inn. In effect, we find that the resolution desired forces us to compromise. The more detail we want the less area that we can see. Keep this concept in mind, as similar trade-offs are true for audio systems.

Photography

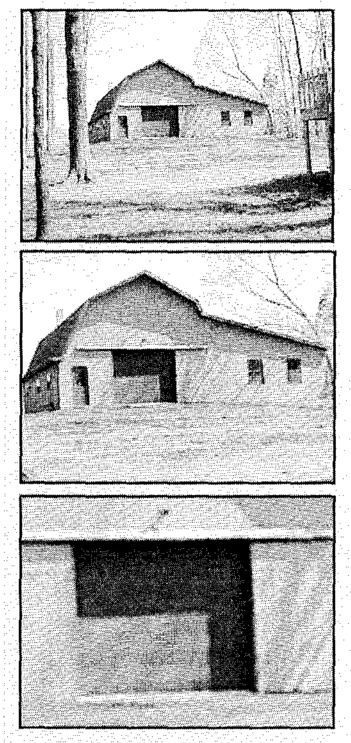
Resolution is also a factor when taking photographs. A zoom lens allows the photographer to select the area to be photographed. The more area that is included in the picture, the lower the detail that can be resolved about that area. Again, it is an appropriate resolution that is desired.

Digital Photography

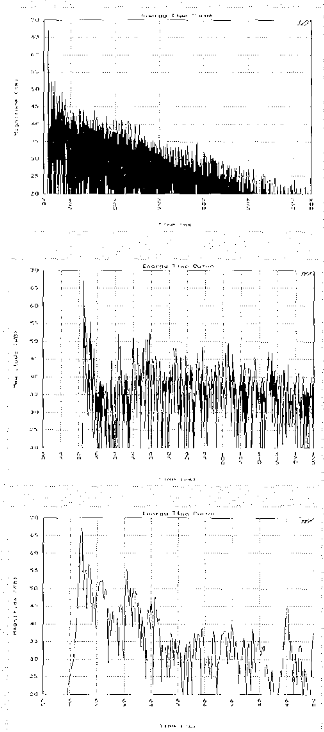
With the advent of digital cameras, we now must choose the resolution desired when making photographs. Just like an audio signal, an image is "digitized" by dividing it into small sections, with a numerical value being assigned to each. In a "gray scale" photograph, this value is typically between 0 and 256, each being a different level of gray ranging from black to white. 8-bit, or 256 levels of gray is sufficient, since this value surpasses human perception of grey scales.

Since digitizing, whether in audio or photography, involves breaking something big up into small pieces, it must be decided how large each piece will be. This will ultimately determine how much detail can be resolved about the object. How big is each section of the photograph? It depends upon the resolution desired. Below is an image with 50 dpi (dots-per-inch) resolution. Note that the detail is low and the picture is a poor facsimile of the subject. As the resolution is increased, the picture gets clearer. As with the other examples, there is a point of diminishing returns that is reached, where greater resolution does not provide a better photograph. To increase the resolution beyond this point only results in bigger and bulkier data files.

In digital audio systems, as in digital photography, the audio signal must be divided into small sections, with each being assigned a numerical value. In 16-bit digitization, these numerical values will range from 1 to about 65536. This determines how accurately amplitude levels of the audio signal can be resolved, as well as the minimum value that can be assigned to the noise in the signal. Each time a bit is added, there will exist twice as many values that the audio signal level can take on. Some systems use more bits to achieve improved signal-to-noise ratios in the recorded signal. The other issue is the time quantization of the audio signal. The audio signal must be sampled many times per second during the digitization process, and the more times it is sampled the more precisely it can be recorded. For pure tones, which are the building blocks of all audio waveforms, it is necessary to sample it at least twice to find its rate of change with respect to time (frequency). As such, the required sampling frequency is twice the highest frequency present. Since humans do not hear above 20 kHz, the required sampling frequency for high-fidelity audio is at least 20 kHz x 2 or 40 kHz. In practice, most systems use 44.1 kHz or 48 kHz.



Increasing Focal Length

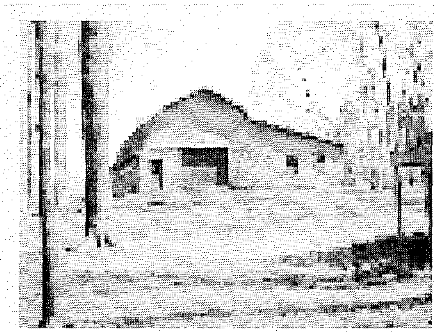


Increasing Time Resolution

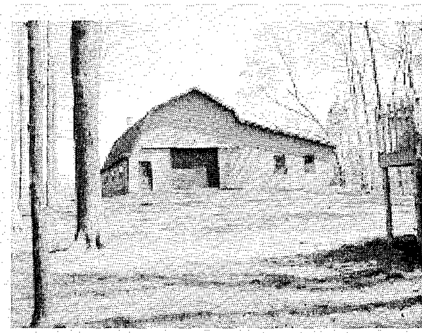
Increasing the time or frequency resolution of an audio measurement is like zooming in with a camera. The higher the resolution, the more detail that can be observed. High time resolutions are useful for performing loudspeaker synchronization, while high frequency resolutions (low time resolutions) are useful for looking at the energy decay in a room. Modern analyzers provide both, allowing the operator to optimize their observation of a system's performance. Like the map example, you must decide the proper compromise in resolution.

$$\text{Time} \times \text{Frequency} = 1$$

$$T = \frac{1}{F} \qquad F = \frac{1}{T}$$

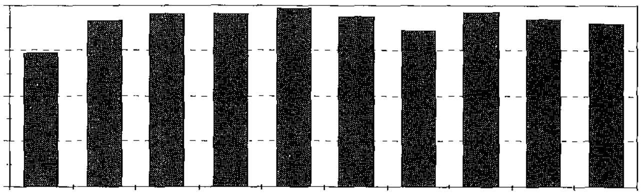


50 dpi resolution

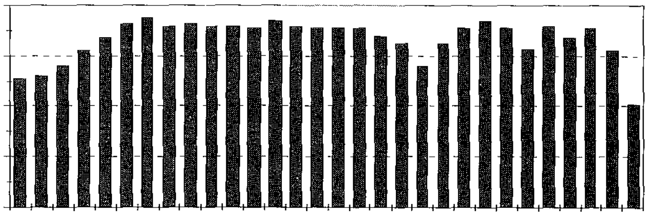


100 dpi resolution

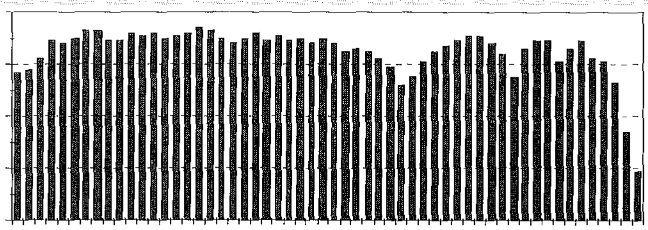
300 dpi resolution



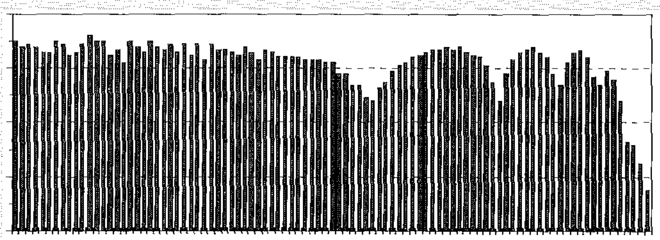
10 data points (one octave)



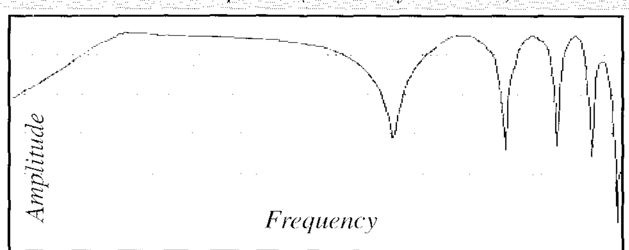
32 data points (one-third octave)



64 data points (one-sixth octave)



128 data points (one-twelfth octave)



1024 data points

How Much is Enough?

The above diagrams show the same data displayed at some common resolutions. As frequency resolution increases, the signal can be viewed with more detail in the frequency domain. The trade-off is that the measurement must cover a longer period of time (lower frequencies last longer than higher frequencies). As such, time resolution must be sacrificed to improve frequency resolution, and vice versa. Notice that interference effects become quite apparent at higher resolutions. Conventional real-time analyzers provide frequency resolutions up to one-third octave. FFT and TDS analyzers can provide much higher resolutions.

Resolution in Audio Systems

There are many choices concerning resolution when designing and measuring audio systems. One common example is in equalizer types. A mixer may have a bass and treble control on each channel for changing the "tone" of a program source. The master section may have a ten-band equalizer for changing the spectral balance of the overall mix. Equalizers are readily available in one-octave, two-thirds octave and one-third octave resolutions, each providing higher resolution control of the audio signal. The most precise equalizers that we use are parametrics, which can provide filters that are only a fraction of an octave wide. As with our other examples, the equalizer's resolution must be appropriate for the task at hand.

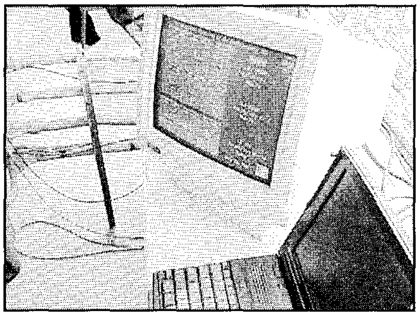
It was once thought that one-third octave analyzers provided the highest resolution needed in audio system work, since notches in response that are narrower that this aren't all that perceptible to humans. On the contrary, we find that the higher resolution analyzers allow us to observe and diagnose problems that are simply not visible on one-third octave analyzers. Let's look at why this is true.

Just as in the example using the road map, the chosen resolution determines what can be observed. There is a trade-off between detail and the overall picture. In audio system work, the resolution trade-off is between the time and frequency domains. If more information is desired about the frequency response of the system, resolution in the time domain must be sacrificed. The opposite is also true. As such, there is a "tug-of-war" between time and frequency resolution.

For instance, it may be desirous to observe the total decay of energy in a room on an analyzer screen, perhaps to measure the reverberation time of the space. If the room's decay were on the order of 3 seconds, this would be the required setting of the horizontal axis of the display. In order to achieve this "coarse" 3 second time display with an FFT analyzer, a very low time resolution (or, very fine frequency resolution) is required. The opposite would be true if we desired a very precise time resolution to observe the difference in the arrival time of a woofer and tweeter in a two-way system. This high precision in the time domain comes with the sacrifice of resolution in the frequency domain. Again, the correct resolution is the one that is appropriate for the task at hand.

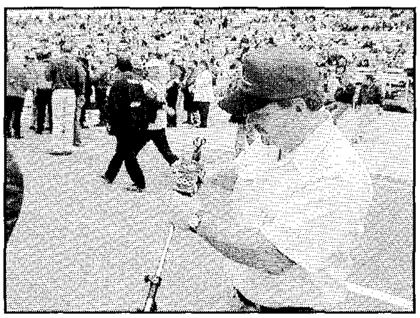
The issue of resolution takes on even more importance as digital sampling and processing play an increasing role in sound reinforcement. Time quantization determines the highest frequency that a system can reproduce, and amplitude quantization determines the maximum signal-to-noise ratio of the system. While current performance levels yield acceptable results, faster and more powerful processors will eventually make digital signals indistinguishable from their analog counterparts in every detail. Anyone want to buy a turntable?

"Gentlemen... start your engines!"

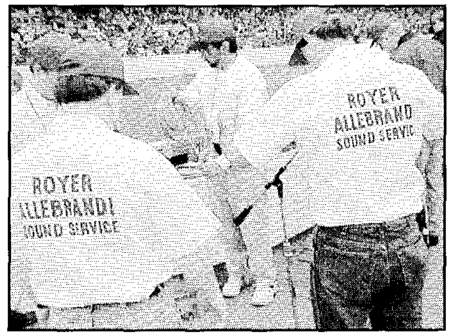


The Indy 500 represents one of the most technologically advanced sporting events on the planet. With a seated audience capacity that more than doubles it's next closest rival, the technology involved in providing sound reinforcement on race day is a unique mixture of modern and tried-and-true techniques.

Over 60,000 Watts of amplifier power is required to deliver the public address messages to hundreds of loudspeakers separated by many miles of cable. Even 70.7 Volt distribution would have excessive line loss in this facility, so the distribution voltage was raised to a hefty 240 Volts by proprietary step-up transformers on each of six 10,000 Watt Crown Macro-Tech amplifiers. The amplifiers are monitored by the Crown IQ System, which alerts the operator of any problems that may be developing.



One of the most important audio passages on race day is the single historical line uttered annually by Mrs. Hulmann, "Gentlemen, start your engines!" A crew of five (or more) is involved in setting up a Crown Tridundant microphone, which is actually three mics in one. The event is the culmination of hours of pre-race activity, and the focal point of every eye and ear in the facility, as well as millions upon millions of watchers and listeners worldwide. Once the proclamation is complete, the audio personnel breath a sigh of relief, and retreat to the tower to watch the race. The equipment is whisked away to the equipment room, where it will remain until the next big race, which is the Brickyard 400 in August, when the whole scenario will be repeated.



On race day, the Indianapolis Motor Speedway is a boiling pot of technology, with miles of fiber optic cable, dozens of satellite uplinks, and a sea of telemetry and wireless communication devices. With all of the technology at hand, John Royer chooses a proven method for signaling the correct time for Mrs. Hulmann's proclamation. With all eyes looking on, the signal is given and the race begins. If only the rest could be so simple!

An Economical **Redundant Processor** Selector

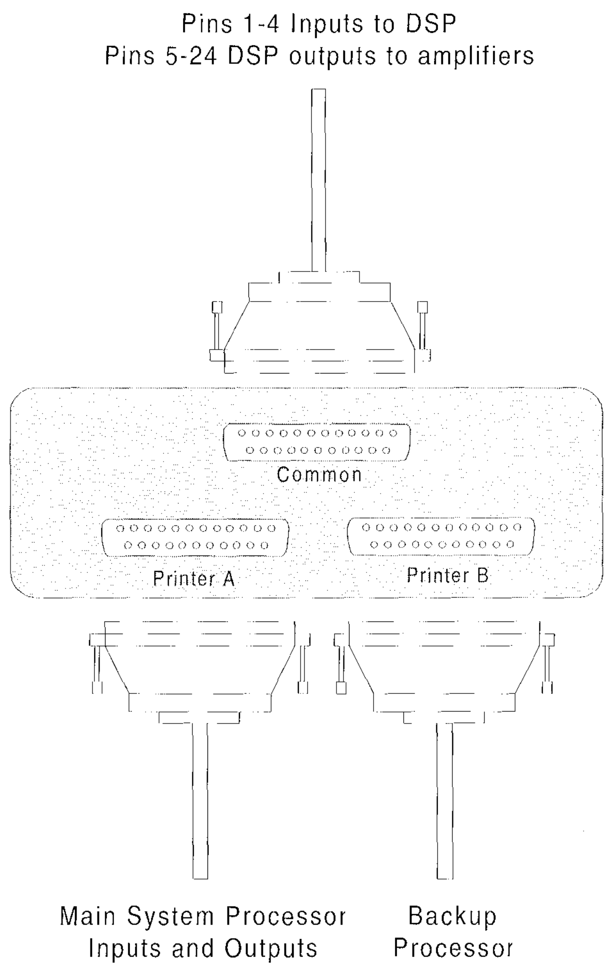
Digital Signal Processors, better known as DSP's, represent the most important development in audio in recent years. They are now affordable, reliable, and a single unit can replace an entire rack of analog processing equipment. The trend these days is to place one or more of these units at the heart of the sound system to provide equalization, crossover and delay functions.

In spite of their advantages, even the best DSP processors can and do fail, often rendering the entire system inoperative. The only solution to the problem is redundancy, which means that a similar unit (or analog equivalent) must be available as a backup. Switching to the backup unit in the event of a failure can be cumbersome, especially if there are not trained personnel on hand to accomplish the task. Here is a simple, quick, and affordable way to switch to a backup processor "on-the-fly."

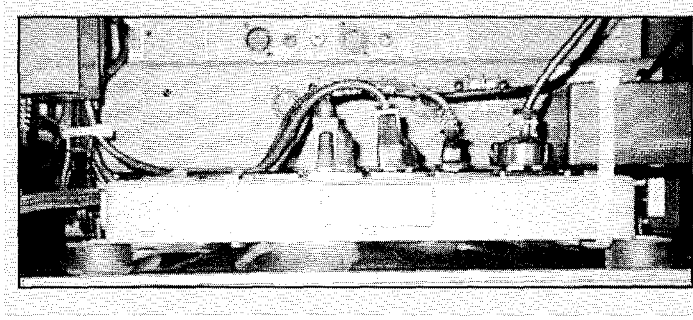
The method involves using a computer printer selector switch to toggle between two processors. These units can switch 25 contacts simultaneously, which accommodates up to 12 wire pairs, just the right amount to switch all of the inputs and outputs of a 2-in-4-out processor. You can custom-make the cables using raw DB25 connectors. If the cable runs are short, it is even possible to fabricate the cables from parallel printer cables, which eliminates the need to wire-up one end. The absence of a shield on these short runs is not a problem in many applications. It is only necessary to switch the signal conductors (pins 2 and 3). The shields (if present) can be connected at the equipment and disconnected at the switch.

Such switches are readily available for less than $50, and can easily be replaced if they go bad (very unusual). Since they are available from any computer or office store, they offer an attractive alternative to expensive, proprietary switching configurations.

A prominent Midwest arena has been "saved by the switch" three times since the installation of a new system three years ago, averting a major audio disaster (full-house of paying customers with no sound system!) each time.



Wall Warts and Line Lumps...



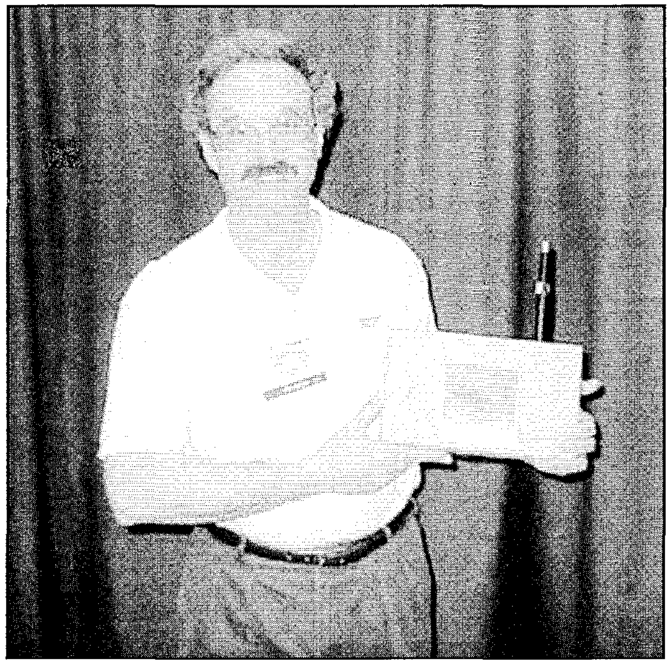
External AC and DC power supplies are something that we must learn to live with. They offer some advantages in that better noise figures are obtainable since the AC line voltage is isolated from the circuitry, not to mention the fact that the use of these devices makes UL approval easier to get, and therefore reduces the cost of the device.

A good way to mount these devices in a rack is to use a couple of rubber feet as standoffs for the power strip. They serve as a sort of "shock absorber" for a wall wart, as well as providing a way to strap it down with a large cable tie.

Farrel Becker and the TEF-PAD

The engineers at Crown International have been busy creating the next evolutionary step for the TEF analyzer; the TEF-PAD (Portable Acoustic Device). The TEF-PAD is a hand-held device that is capable of performing a variety of audio and acoustic measurements. It's numerous features include a PCMCIA card slot that can used for disk storage, as well as a serial port for communication with a host PC. The integrated LCD display allows the user to roam freely with the device, with all user controls being in the form of a touch screen on the display.

Crown sees the TEF-PAD as an enhancement for the TEF 20, not a replacement. The first application for the TEF-PAD will be a real-time analyzer. The capabilities of the device will be enhanced in the coming months as new software becomes available.

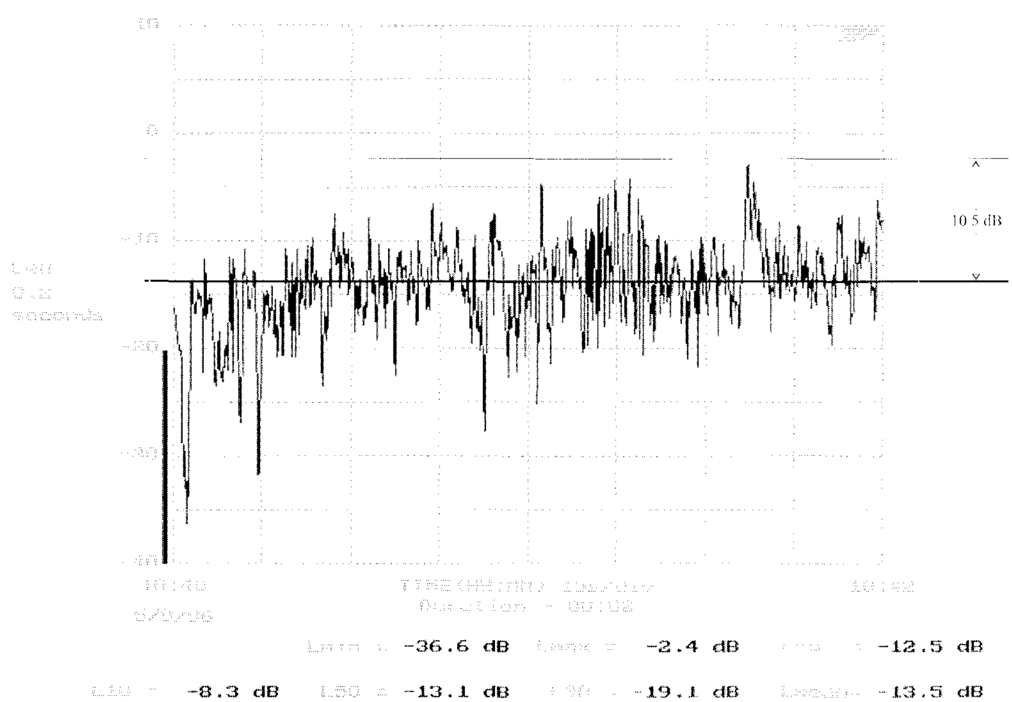


Farrel proudly displays the TEF-PAD

How Much Headroom?

All program material has an average level that is perceived by the listener (and the sound level meter) as the loudness of the system. Unless you only play sine waves through your system, some headroom above this average level is required to pass the short transient peaks that characterize speech and music.

The question of required headroom in a sound system always stirs up some debates, primarily due to the great variety of program sources that must be reinforced by the system. In our own testing and experience, we find that a minimum of 10 dB is required, with a maximum practical value of 20 dB. It is quite easy to have 20 dB of headroom through the signal processing stages, but the bill comes due at the power amplifier, where 20 dB of headroom means that there must be 100 times the average power used available to pass peaks in the material. In the real world a limiter is often used ahead of the power amplifier to reduce the needed headroom to an affordable level. The figure below shows the amplitude of a popular recording as a function of time. Note that for this material (quite percussive) 10 dB of headroom would be sufficient. In the case of a live performance without excessive limiting or compression, more headroom would be required. It doesn't hurt to have too much, so it is often the budget that determines system headroom.



Cluster Lobe

A long awaited module enhances the EASE program

Effective array design has always been difficult to execute at the drawing board stage of a project, due to the complexities involved with pattern overlap and varying acoustic centers and origins. Much light was shed on the subject at last year's Horns II Workshop, where Mark Ureda gave a memorable presentation on some computer modeling techniques for complex arrays. We have anxiously awaited the emergence of this and similar work in a form accessible to the working audio professional. The latest addition to the EASE program represents this emergence, and promises to be a valuable tool for array design.

Cluster Lobe allows the coverage of a multi-loudspeaker array to be modeled in advance of actual construction of the array. The system design is carried out in EASE in the usual manner, with the room and loudspeaker system constructed using an X,Y,Z coordinate system. The user can then enter the Cluster Lobe module and select the individual loudspeakers that will make up an array. Once the individual loudspeakers are selected, a variety of coverage patterns can be generated, as the accompanying diagrams from the program depict. It is possible for the user to try various spacings, degrees of overlap, and delay times and see how the coverage of the array will be affected. Individual octave bands can be selected, which allows various types of arrays to be constructed, from low-frequency line arrays to voice-range sound columns.

A subject of this complexity warrants several viewpoints of what might happen when these arrays are actually constructed. Dr. Wolfgang Ahnert, co-developer of the program, has included the work of Mark Ureda in the module, as well as other algorithms that he and co-developer Rainer Feistel have developed.

Cluster Lobe represents a significant evolutionary step in the EASE/EARS programs, and hopefully one that will lead to an even more widespread use of this powerful design tool. Cluster Lobe will be demonstrated at the Syn-Aud-Con Advanced Design seminar this summer in Indiana.

The diagrams on the next page were reprinted from the EASE program using the Cluster Lobe module. Modeled is the construction of a low-frequency line array that can offer some directivity in the problematic 250 Hz octave band. The individual components are 15" loudspeakers in enclosures. Since the 250 Hz octave band is being considered, the tuning of the enclosure is not a prime consideration.

Figure 1a represents the vertical and horizontal coverage of a single enclosure. Note that the 0 degrees axis is oriented toward the top of the page for the polar, while figure 1b orientates the listener as looking straight into the diagram. The polar pattern is consistent with what one would intuitively expect from a front-loaded design.

Figures 2a and 2b represents the effect of adding another, similar device stacked on top of the previous one. Note the narrowing of the vertical coverage, while the horizontal coverage remains unaffected. Of interest are the angles of cancellation at the plus and minus 90 degree angles. A microphone positioned here would exhibit good gain-before-feedback in the low-mid region.

Figures 3a and 3b depict three such devices stacked. The on-axis coverage continues to narrow, while the off-axis coverage exhibits lobes that could make mic placement critical. In this configuration, the vertical coverage is less than 50 degrees, pretty good directivity for this octave band.

Finally, figures 4a and 4b represent a stack of 4 devices. The very narrow vertical coverage angle would make this ideal for a large arena or reverberant cathedral. We have often encountered horizontal bass arrays such as this, where an accidental line source was created that ruined bass coverage. Such arrays should be stacked vertically to maintain the horizontal coverage.

Cluster Lobe will certainly become an important tool for the serious system designer, and we look forward to it's continuing development.

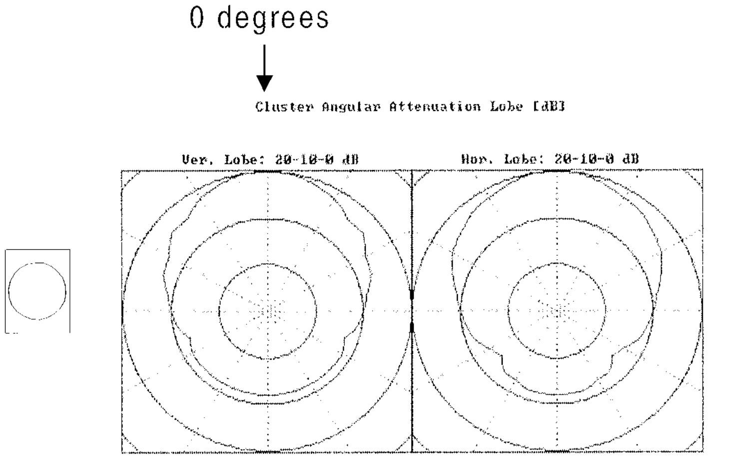


Fig 1a

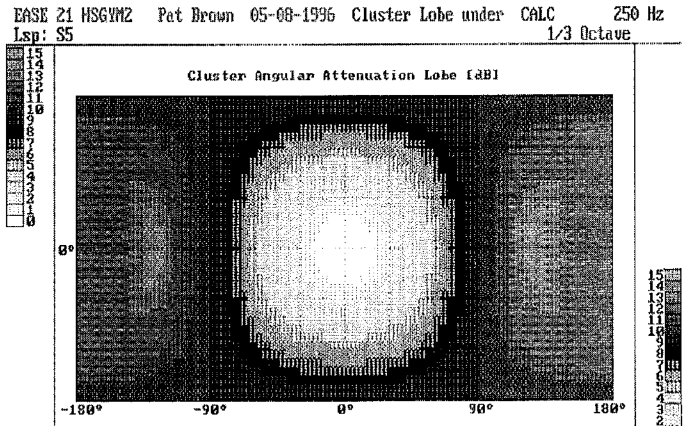


Fig 1b

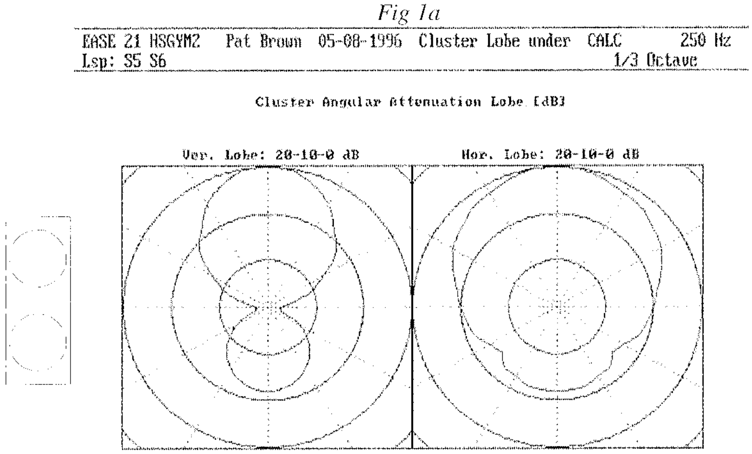


Fig 2a

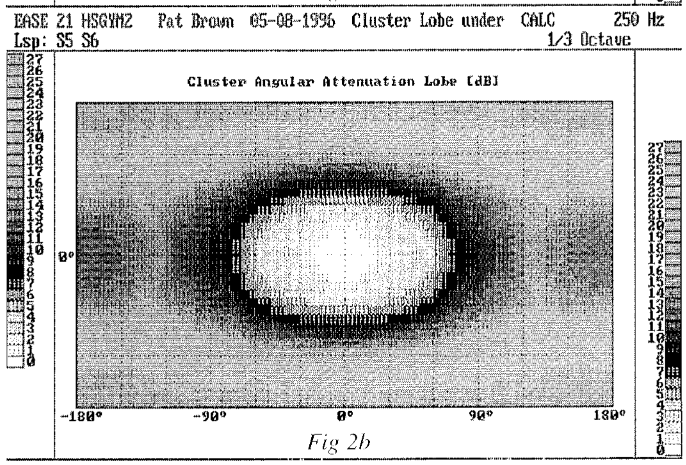


Fig 2b

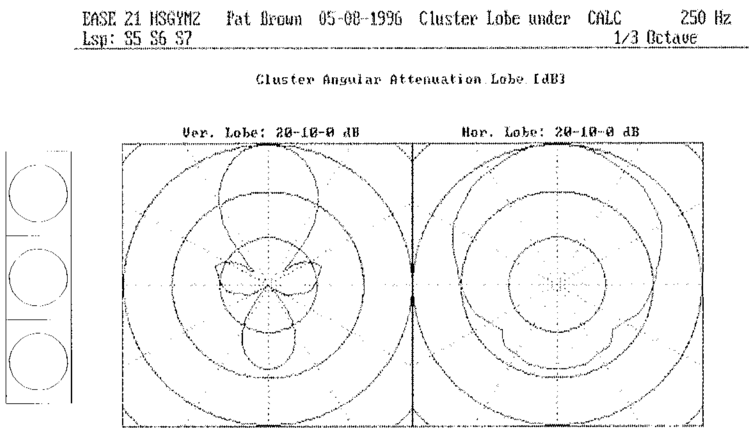


Fig 3a

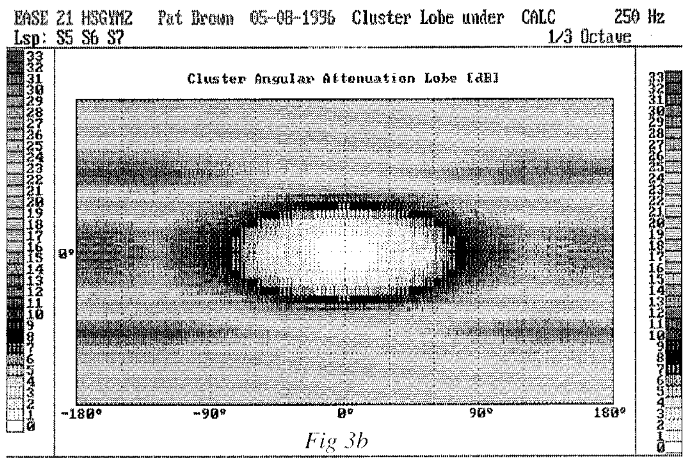


Fig 3b

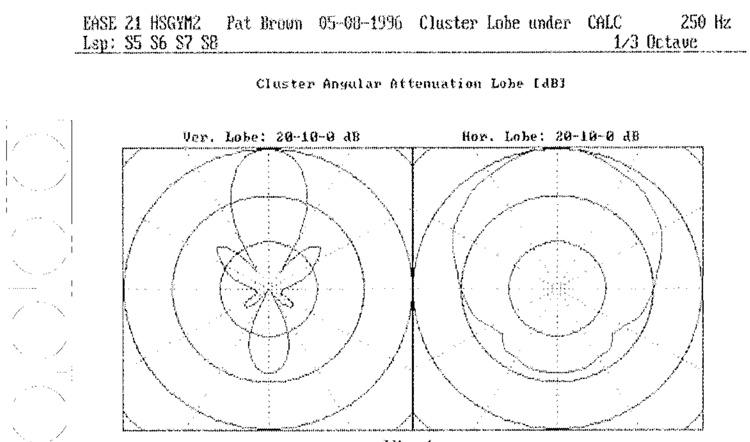


Fig 4a

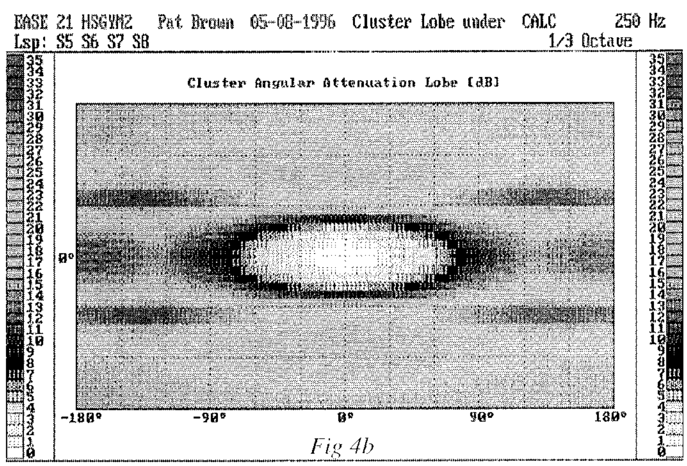
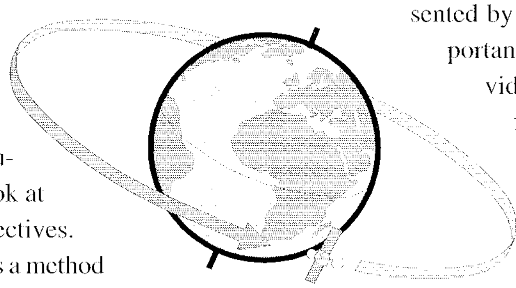


Fig 4b

Cycles, Circles and Triangles

Part III - An Introduction to the Fourier Transform

In our last issue this series dealt with one of the most fundamental audio measurements; the impulse response. You may recall that in it's simplest form, the impulse response is a time domain description of the sound as it passes the microphone, and represents the potential energy present in the sound wave. Even though it represents but one view of the energy present, other perspectives are possible and can be obtained by post-processing the impulse response. For instance, the Hilbert Transform represents one of several ways to determine the kinetic energy component from the potential by shifting the observation point of the signal by 90 degrees. Transforms are useful in audio because they provide a way to look at the same thing from different perspectives. This installment of our series describes a method of transforming time domain data into frequency domain data, which can provide some enlightening viewpoints that might otherwise be overlooked.



Jean Baptiste Joseph Fourier was an 18th century French mathematician, who, like other notables from history, observed and quantified a simple truth about nature. The best way to solve any problem is to break it down into simpler, individually solvable problems. Fourier showed us that complex waveforms can be broken down into individual sinusoids, which are in effect the building blocks of all complex waveforms. You may recall from the first part of this series that a sine wave can be described in the time domain or the frequency domain (inverse-time), and that each of these represents an alternate view of the same thing.

Figure 1 shows a sine wave in the time domain. Note that it is a continuous, ongoing event, analogous to the rotation of a wheel or the hands on a clock. Figure 2 represents a view of the same event in the frequency domain. Note that it is now represented by a single spike, rather than an event that is spread out. The Fourier Transform allows a complex time domain waveform to be described in the domain of inverse time, or frequency. It is in effect a global (continuous, ongoing) to local (represented by a single line) map. Of fundamental importance is that the Fourier Transform only provides a way to observe the event from a different perspective. It does not change or alter the event in any way, and does not add to or remove information from the event.

Let us use an example from everyday life. Consider the rotation of the earth around the sun. Since mankind has only observed and recorded this phenomenon for a relatively short period, we can only approximate the frequency of rotation, which is about once every 365 days. During this annual orbit, the earth itself rotates once each day. Both of these are continuous, simultaneous ongoing events which we have described in the time domain. For a more resolute description of these events, we can break them down into smaller and smaller components. Consider the hour hand on a 24-hour clock, which completes one orbit per day. The minute hand in turn completes 24 rotations, and the second hand completes 1440 rotations each day. To establish a common reference for all of these events, we will use one 24 hour

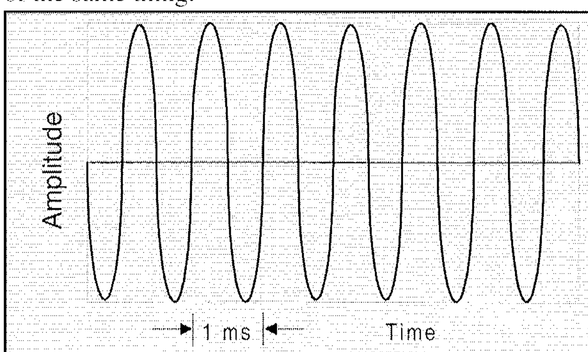


Figure 1

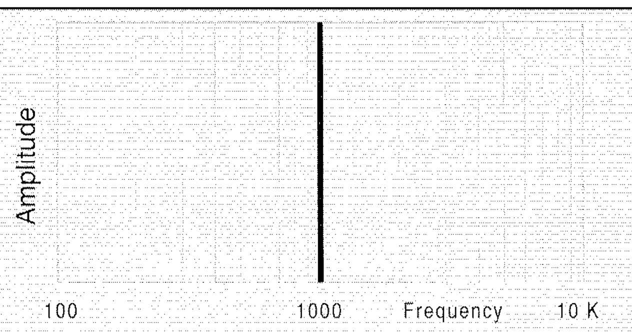


Figure 2

day. All events can then be described in either time of duration or cycles per day. Figure 3 shows each of these events described in the time and frequency domains.

Notice that either time or frequency can be used to describe the same event, and that the "complex" rotation of the earth around the sun can be broken down into individual rotation components. In effect, the complex waveform is comprised of simpler, individual rotations. Also apparent is the fact that these events cover a very wide range of both time and frequency, so much so that it would be difficult to plot them on a linear graph.

The realm of audio and acoustics bears many similarities to this example. The complex waveforms of speech and music can be observed from a perspective of either time or frequency, each being a legitimate alternative view of the event. The Fourier Transform provides a map between these two perspectives, so that knowing one, the other may be determined.

Computers execute a computationally efficient version of the Fourier Transform, known as the Fast Fourier Transform, of FFT.

Event	Period	Frequency
Year	365 days	0.27 cpd
Day	1 days	1 cpd
Hour	0.041667 days	24 cpd
Minute	0.000694 days	1440 cpd
Second	0.000011574 days	86,402 cpd

cpd = cycles per day

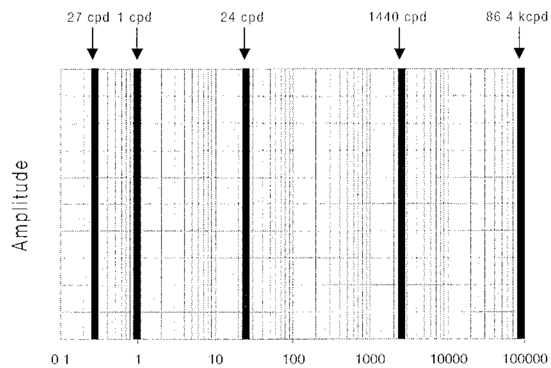
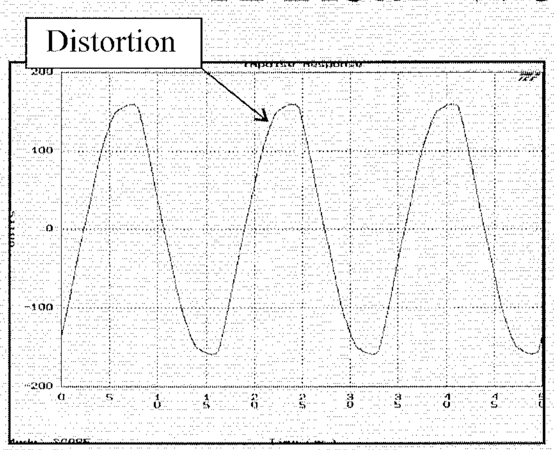
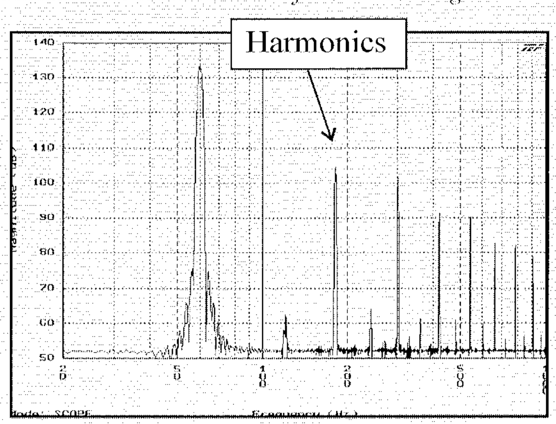


Figure 3

A Real-World FFT Example



Time domain view of AC line voltage



Frequency domain view of AC line voltage

The top figure to the left is the sampled output of an AC outlet at a hotel where we recently held a class. It was captured using a TEF 20 analyzer. Of course, since the voltage is around 110 Volts, it was necessary to first transform the voltage down to an amplitude that the TEF can accept without overload (around 1 Volt). The voltage was sampled using the scope mode of the MLS software package, which is a useful tool for capturing and analyzing audio events.

The waveform has the general appearance of a 60 Hz sine wave, but note that there is some deformation from what a pure tone would look like. The next step was to observe the waveform in the frequency domain, which was accomplished by performing an FFT (Fast Fourier Transform). While this sounds like a complex procedure, it is actually accomplished with the push of a button and a wait of a few seconds.

The frequency domain view of the waveform reveals that the deformation is due to a multitude of harmonics on the line. These could have been generated by something plugged into that particular circuit, such as a computer or an amplifier with a switching power supply. The presence of such harmonics can present problems for certain types of equipment, so this is a good way to test the purity of the AC supply of a venue.

The point is that what may not be obvious in one domain (time, in this case) may be very obvious in another domain (in this case, frequency). Today's serious analysis tools allow the user access to these domains and others. It's a fun age to live in, at least concerning the realm of audio.

A Simple Example of Fourier Analysis

Even though we have been talking about planets and clocks, the exact same principles apply to audio and acoustics. Remember that what we are talking about here are things that rotate, or move in cycles, which seems to be a fundamental and natural form of movement in the world around us. Since the sine wave describes a pure, cyclical motion, the same principles will hold true for anything that moves in cycles, whether it be the orbit of a planet, the hands of a clock, a wheel on a car, or a loudspeaker diaphragm. Figure 1 depicts an audio waveform with a period (time required for one complete cycle) of 0.01 seconds (time domain description) or a frequency of 100 Hz (frequency domain description). Figures 1-3 represent signals with periods of 0.01 s, 0.005 s, and 0.002 s, respectively, or frequencies of 100, 200, and 500 Hz. Figure 4 represents a

Time and frequency are but two different ways to describe the same event...

complex waveform, the combination of all of the previous waves. According to Fourier, even though the waves have been combined, they still retain their individual identities, and through an appropriate playback system a listener would be able to distinguish each individual tone. The Fourier transform is the process of taking this complex waveform, breaking it down into it's individual discrete tones and displaying the result in the frequency domain. Figure 5 is a frequency domain description of the complex waveform of figure 4. We have used the JBL Smaart program to execute the Fourier Transform to get from one to the other. Real-world audio waveforms are very complex, and require microprocessors to sample a waveform in the time domain, perform the FFT, and display a spectrum in the frequency domain.

As you can see, time and frequency are but two different ways to describe the same event, and the Fourier transform simply provides a way to get from one description to the other. Next time we will deal with the actual mechanics of the process and some of it's side effects, and gain some more insight into this powerful analytical tool.

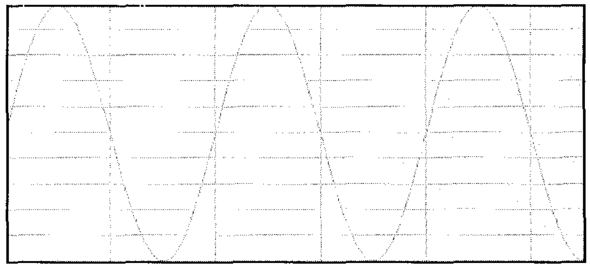


Figure 1 - 100 Hz sine wave

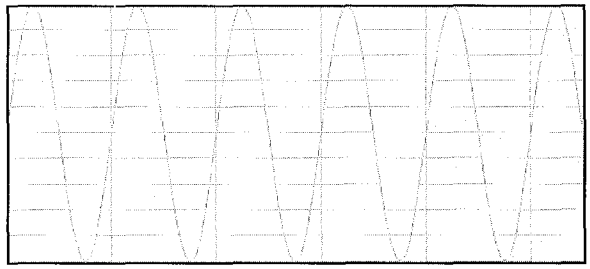


Figure 2 - 200 Hz sine wave

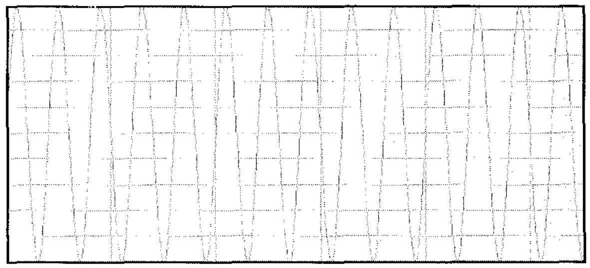


Figure 3 - 500 Hz sine wave

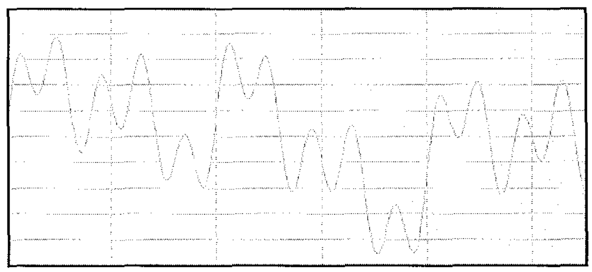


Figure 4 - Combination of all three waveforms

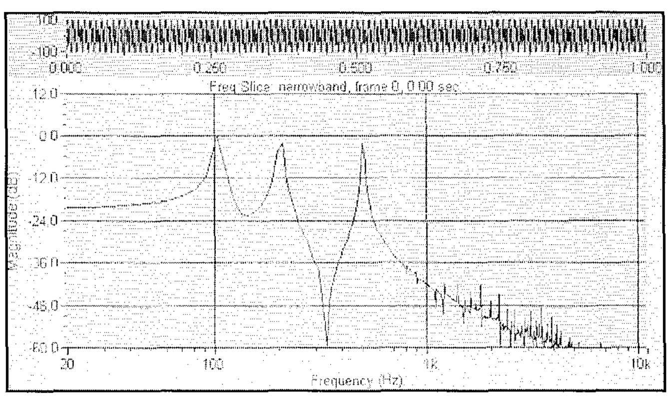
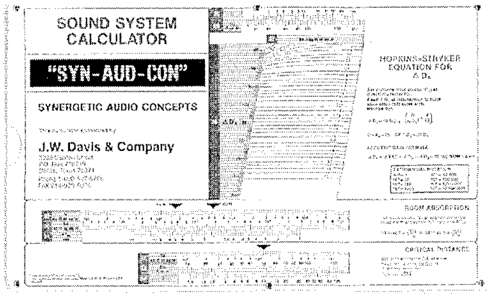


Figure 5 - The complex wave of figure 4 after the fft (Fast Fourier Transform) has been performed. Note the three distinct bands of energy and their spectral centers.

Finding the Equivalent Acoustic Distance (EAD)



The Equivalent Acoustic Distance (EAD) of a space is a system design parameter that is dependent upon the level of the talker and the level of the noise in the space. It represents the distance from talker to listener that communication can take place with a normal voice and be unhindered by the noise in the environment. The goal of the system design is to place every listener (by virtue of the sound system) at the Equivalent Acoustic Distance.

To determine this important design parameter, it is necessary to first establish some points of reference. The first is the level of the talker. Of course, this varies for different individuals, but the generally agreed upon levels for talkers are as follows:

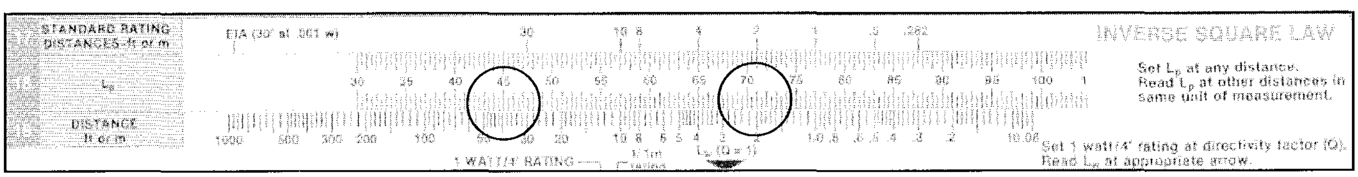
Weak Voice 65 dBA @ 2'
Normal Voice 71 dBA @ 2'
Raised Voice 77 dBA @ 2'

Another parameter is the ambient noise level in the space. This is commonly measured using a sound level meter with an "A" weighting scale. This should be a "worst case" estimate, taking into account HVAC systems, traffic noise, etc.

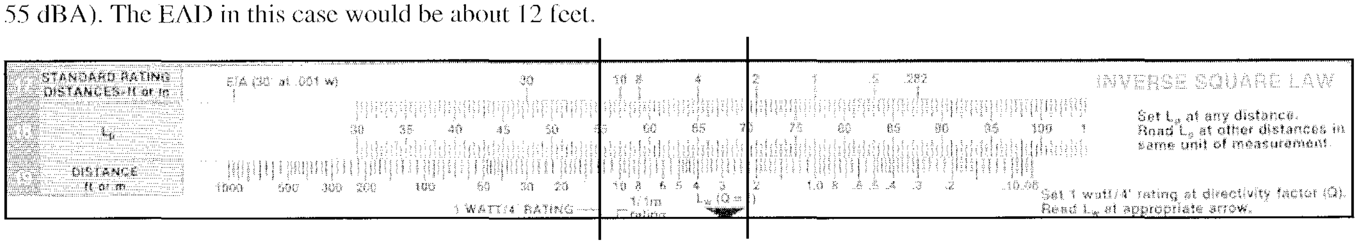
The last parameter is the desired signal-to-noise ratio for the system/room combination. For auditoriums (places for listening) S/N ratios on the order of 25 dB should be attainable. In the real world of high school gymnasiums and sports arenas, 10 dB of signal-to-noise can be difficult to achieve. In short, the more the better, with the ultimate limitation being the sound level from the sound system that the audience is willing to endure. If we were to place a "ceiling" of 100 dBA on the level of the sound system, we would not want the ambient noise to exceed 75 dBA for optimum communicating conditions.

Example: Determine the EAD for a normal talker in an auditorium with an ambient noise level of 45 dBA. The desired signal-to-noise ratio is 25 dB.

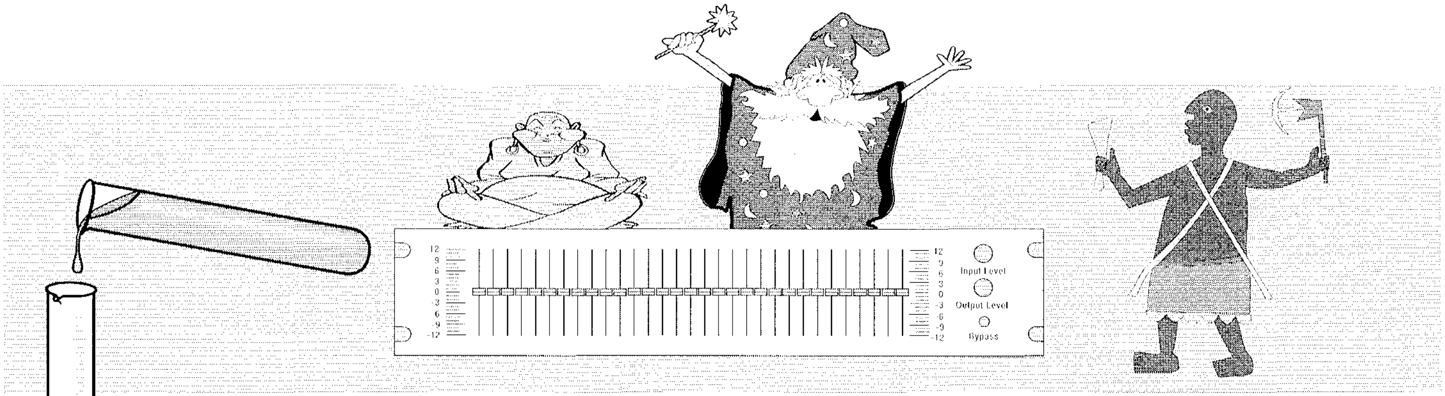
Solution: Since the level of a normal talker is 71 dBA at 2', align this relationship between scales 18 and 19 of the Inverse-Square Law section of the slide rule. Once this relationship is established, observe that at a distance of 40', the level of the talker would equal the level of the room noise (S/N ratio = 0 dB). To calculate the EAD for a 25 dB S/N ratio, add 25 dB to the room noise (45 dBA + 25 dB = 70 dBA). Observe the physical distance on Scale 19 below the 70 dB mark on Scale 18 for an EAD of 2.2 feet.



If the desired S/N ratio were only 10 dB, read the distance on scale 19 opposite 55 dB on scale 18 (since 45 dBA + 10 dB = 55 dBA). The EAD in this case would be about 12 feet.



EAD's for 10 dB of S/N and 25 dB of S/N, respectively.



The Art and Science of Equalization

Few subjects generate as much controversy as equalization. The methods currently in use range from the purely scientific to the mumbling of incantations and burning of incense. Having heard (and tried) about all of them, I offer my "two cents" worth.

It should be remembered that what the listener hears is actually a combination of responses from several "systems" each providing its own unique contribution to the sound. It must also be pointed out that these various responses are spread out over time, making the listening experience even more complex and elusive of definition. Let us look at each "ingredient" in the recipe of listening, before determining when and if it may be desirable to attempt to equalize them.

The word "equalize" means to bring to equilibrium. The electronic equalizer can add to or remove energy from the electrical signal (ahead of the power amplifier) in a frequency-dependent manner. This is typically done in an attempt to compensate for the behavior of something further up the signal chain. Fundamental to the concept of equalization is the fact that only the electrical signal ahead of the power amplifier is being directly manipulated by the equalizer. All other systems after the equalizer are not being changed, only the energy delivered to them! This includes the loudspeaker, room, listener, etc.

The following "ingredients" determine what is heard by a listener in an auditorium.

The Electrical Signal

We will define the electrical signal as that signal which is presented to the loudspeaker's terminals for transduction into an acoustic signal. It is a composite of responses of all devices ahead of the point of introduction to the loudspeaker terminals, which includes talker, microphone, mixer, equalizer, amplifier, wire, etc.

The Loudspeaker Response

Since the loudspeaker has the critical task of converting the electrical signal into an acoustic signal, it adds its own unique signature in the process. Other than the room itself, the sound heard by the listener is probably affected more by the loudspeakers response than any other response in the chain. Since no perfect loudspeakers exist, the only standards that exist for comparison are theoretical ones, which makes rating loudspeakers a very difficult task, and explains why so many shapes, sizes, and brands exist.

The Direct Sound Field

By definition, the direct sound field L_D is that sound which arrives at the listener's ears having not encountered any room surfaces. Outdoors, it is the predominant sound field. Indoors it is the predominant sound field inside of critical distance. L_D is a combination of the responses of all devices up to and in-

Direct Equalization Methods	Adjust filters for desired response	Change or redesign components	Change time/distance relationship between spks	Add absorption or diffusion to wall surface	Increase or decrease Sabins for individual octave bands
Component	The Electronic Equalizer (Graphic or Parametric)	The Loudspeaker Response	The Loudspeaker Array Response	Early-Reflection Response	Reverberation Response
Indirect Equalization Methods	Increase or decrease energy at specific frequencies	No change possible without affecting previous component!!!			

Figure 1 - The hierarchy of responses for a sound reinforcement system

cluding the loudspeaker. This is an important point, as it should be pointed out that corrections introduced to compensate for a loudspeakers response come at the expense of "corrupting" the otherwise correct signal ahead of the power amplifier. Again, the loudspeaker remains physically unchanged. Only the energy delivered to it has been modified.

Adjacent Loudspeakers

The response of L_D will also be affected by the presence of multiple sound sources. The electrical signals driving adjacent loudspeakers are discrete and independent of one another, but are ultimately mixed acoustically in the environment. This mixing of responses greatly modifies the sound heard by the listener by altering the level and direction of sound radiation at each frequency. Depending upon the time relationship between loudspeakers, energy will sum and cancel in a frequency-dependent manner.

Early Reflections

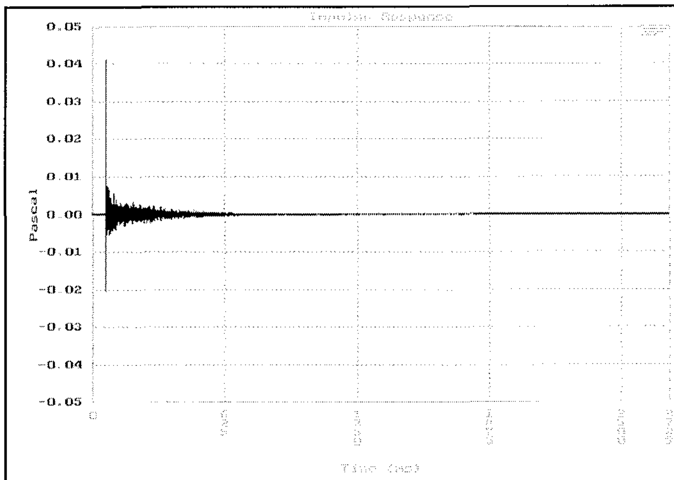
The early-reflected sound field L_{RE} is the room's contribution to the sound heard by a listener. In effect, the room acts like a multitude of sound sources spread out over time and distance. As reflections are returned to a listening position, the ear-brain system combines them with the direct sound field in a frequency and time dependent manner. It is the early-reflected sound field that gives a room its characteristic sound, such as "hard", "soft", "warm", etc. Since the room provides no energy of it's own, the "room sound" arriving at a listening position comes ultimately from either the program source or sound system itself.

Reverberation

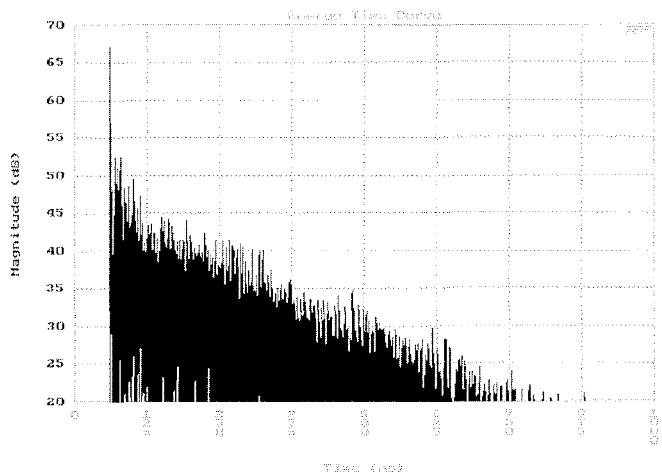
The reverberant sound field L_R results from energy accumulating in the acoustic space. Its contribution to the sound heard by a listener ranges from desirable (choir and organ), to destructive (speech). Since the room is passive (produces no energy of its own) it can only produce reverberation from the energy delivered to it from other sound sources in the room. A hard room is a sort of acoustical "storage tank" that accumulates and releases energy over time.

The Equalization Chain

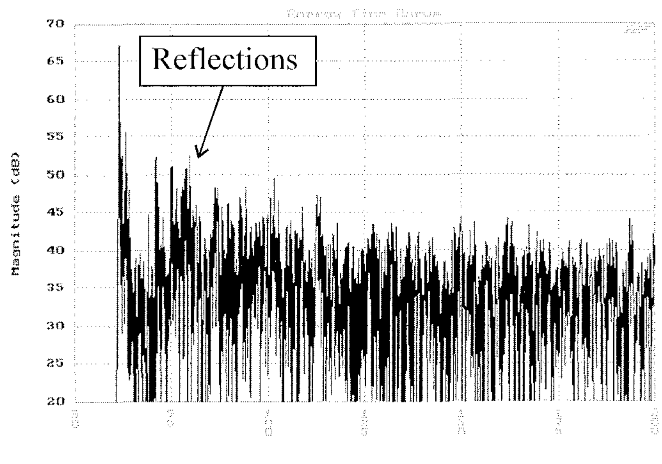
Figure 1 shows the order of "responses" that exist in a sound reinforcement system. No response in the chain can be changed by something that comes later in the chain. For instance, the response of the equalizer cannot be changed by the loudspeaker, nor can the response of the loudspeaker be affected by the room's response. This hierarchy is crucial since much of the "mystery" of equalization involves attempts to violate the order of the chain.



Impulse response of sound system/room combination. Note horizontal time span of almost 3 seconds.



Energy-Time curve display of the impulse response. Since there was no significant energy arriving after 1 second, the time span was truncated for a better display.



ETC with time span truncated to 200 ms. This display shows the direct sound and the early reflected field. It is this span of energy that predominantly characterizes the sound quality of a system/room.

Figure 2

Let us "build" a system response by adding the components individually. This will aid in understanding what an equalizer does and the best way to apply it. Figure 2 is an energy-time curve of a loudspeaker/room combination. It represents all of the energy that arrives at a single microphone position when the sound system is energized with a pulse. We can dissect this response in order to observe the effect of each sound field.

Figure 3a shows the same ETC, with the cursors set to isolate the direct sound field. The direct sound field represents the combination of the equalizer/loudspeaker response. Since this represents the energy that is delivered to the room and ultimately the listener, it should be scrutinized closely for smoothness in response. If the loudspeaker has uniform response over its angles of radiation, any changes made to the on-axis response should also bring improvement at other listening angles. It is good practice to look at the direct sound field first when performing a system equalization, since any interference effects, etc. will be obvious and can be corrected before proceeding with the equalization. Figure 3b represents the spectrum of the time span indicated in Figure 3a. The notch at 500 Hz is probably due to a floor reflection at the mic position.

Figure 4a represents the direct sound field and some early reflections from the room. Note that the "notch" from the floor reflection has been smoothed somewhat by the later energy arrivals. Also note that the mid-high frequency response has not changed much, but that the low-frequency response has increased in level. This illustrates why the sound of a system can change when an audience is present, since the level of the early reflections are affected by the increased absorption. The direct sound field remains unchanged. At this point, the equalizer can compensate for the low-frequency energy build-up by reducing the energy fed to the room by the sound system on a frequency-dependent basis. Remember, you are not correcting the room, only changing the energy delivered to it.

Figures 5a and 5b represent the inclusion of about 160 ms of reflected energy. Note that since these reflections are rather late in time, their effect on the frequency response is very narrow band so the display has been smoothed for clarity. It should also be pointed out that the pattern of reflections (and hence, the frequency response), will change as one moves through the auditorium. Any equalizer adjustments made for one location may be detrimental at other locations. Any equalization to compensate for the contribution of the early-reflected sound field should be subtle and broadband, compensating only for energy build-up (adjusting the power response).

The reverberant sound field is a statistical field, and should be fairly uniform as one moves through the auditorium. Since it represents the frequency-dependent storage properties of the room, it may be useful to make some small compensations for it. Please remember that you are not "equalizing the room", but rather are changing the energy delivered to the room. A properly designed sound system should have uniform power response, so hopefully the energy that the system delivers to the room is relatively the same for the low, mid, and high-frequency sections of the system (see page 26).

A Step-by-Step Guide to Equalization

With all of this in mind, we are ready to draw some conclusions and recommend one possible procedure for system equalization.

Step 1 - Observe the direct sound field at the on-axis position to the loudspeaker. This should be done for each individual loudspeaker, with all others switched off. Any deficiencies in the individual components should show up during this test. If you are using an FFT or TDS analyzer, the time window should include the first energy arrival and the next 10 -20 ms. This will extend the resolution of the display to about 100 Hz, which will be adequate for viewing all but the low-frequency devices.

Step 2 - Repeat the above as additional loudspeakers are switched on. Note any interference effects that dramatically alter the frequency response. If these exist, you may wish to try re-aiming loudspeakers to minimize them, or using precision signal delay to move them. Adjust the equalizer for the flattest possible response over the 300 Hz to 5000 Hz region. What you do to the frequencies outside of this band is largely a matter of taste, and a subject of heated debate among those who like to debate such things. At this point, the purist is finished, since the response of the sound system is optimum. You may realize an improvement by continuing to step 3, but make sure that you can get back to this point if you don't like the further changes.

Step 3 - Increase the size of the time window, or, if using a real-time analyzer move the mic farther from the loudspeaker. Note any changes in the response that occur from the inclusion of more energy from the room. At this point it can be useful to move the mic throughout the loudspeaker's coverage pattern while watching the display to see if any response bumps are common to all mic positions. Reduce the energy at these frequencies with the equalizer and see if the sound improves.

It should be emphasized that listening to program material is an important part of the process. It doesn't matter what the analyzer says if it doesn't sound good. Also, don't wreck things by notching out feedback frequencies to oblivion once you have things sounding good. The rule of thumb: if it takes more than 3 dB to remove a feedback, look for the problem at the source, such as a poor miking technique, an unfortunate mic location, or a sloppy array lobing energy toward the mic.

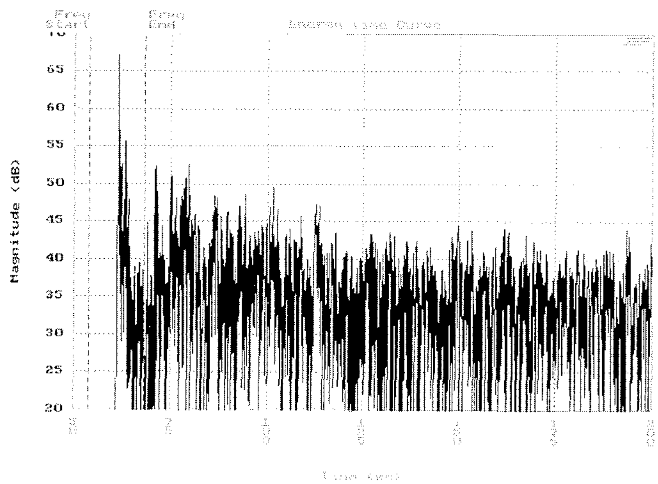


Figure 3a - ETC (note frequency start/end times) of the first 200 ms of energy arrivals at the test mic position.

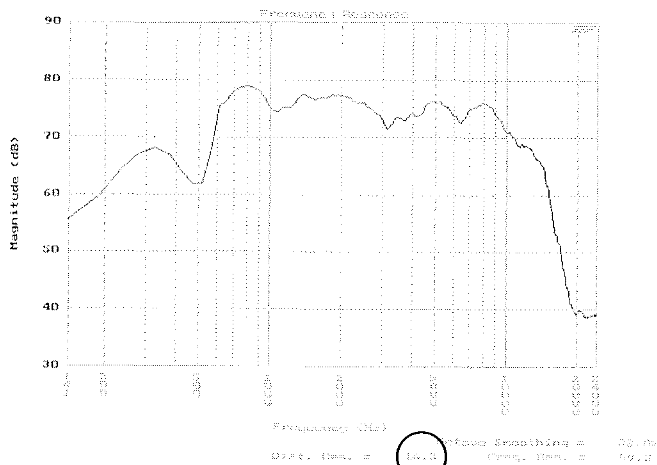


Figure 3b - EFC (energy-frequency curve) of the first 16 ms of energy arrivals at the test mic position. This is essentially the direct sound field.

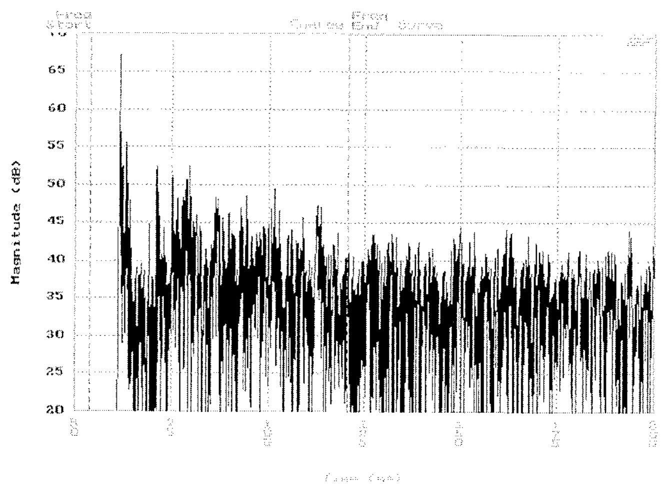


Figure 4a - ETC (note frequency start/end times) of the first 200 ms of energy arrivals at the test mic position.

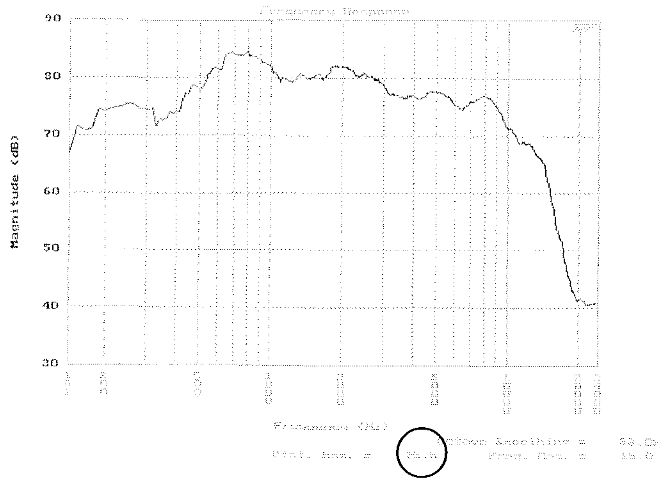


Figure 4b - EFC (energy-frequency curve) of the first 67 ms of energy arrivals at the test mic position. This response includes some early reflections, and has been smoothed to provide a better display.

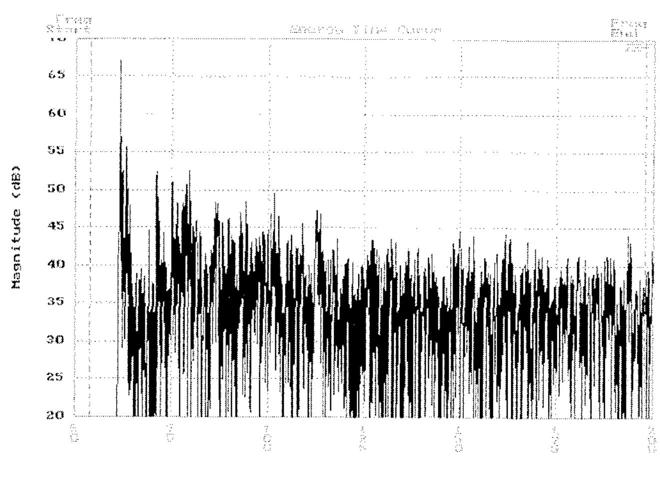


Figure 5a - ETC (note frequency start/end times) of the first 200 ms of energy arrivals at the test mic position.

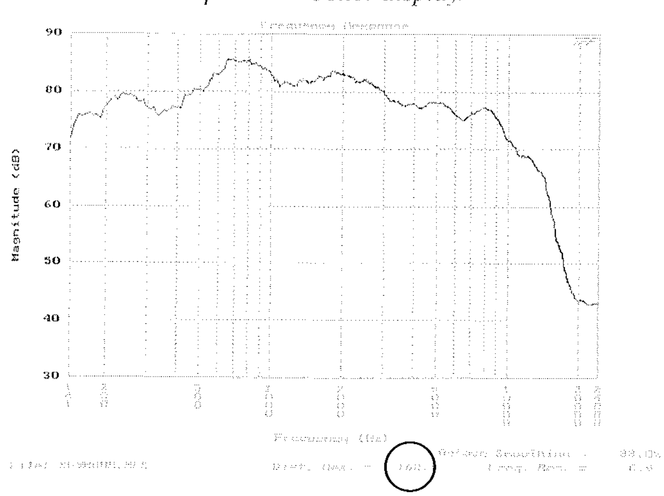
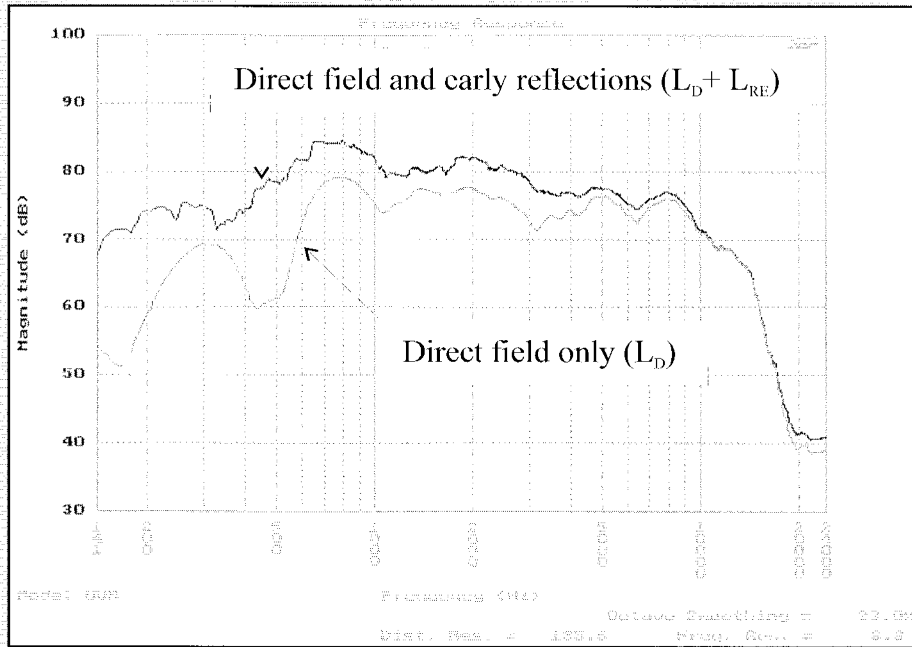
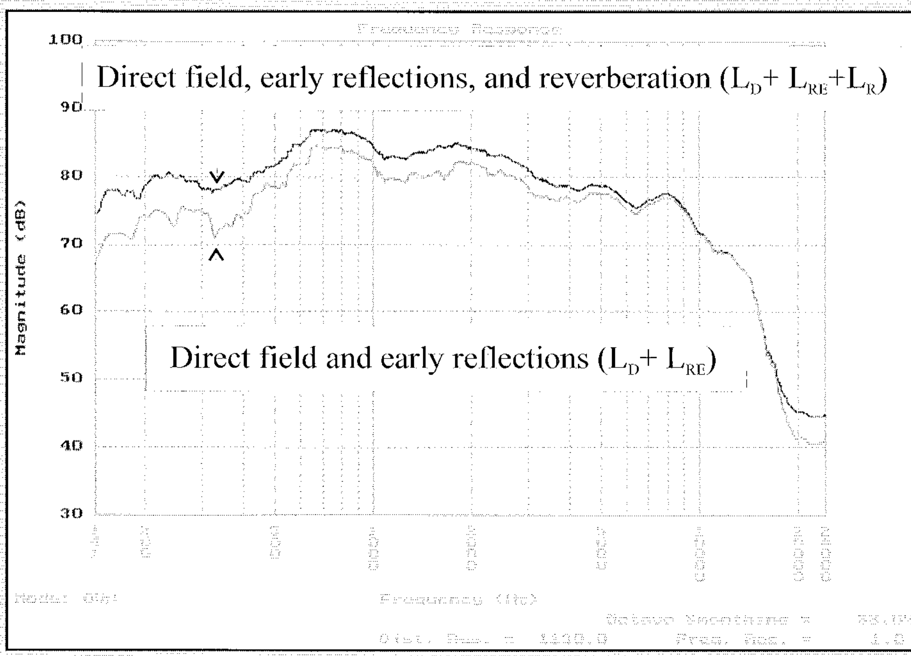


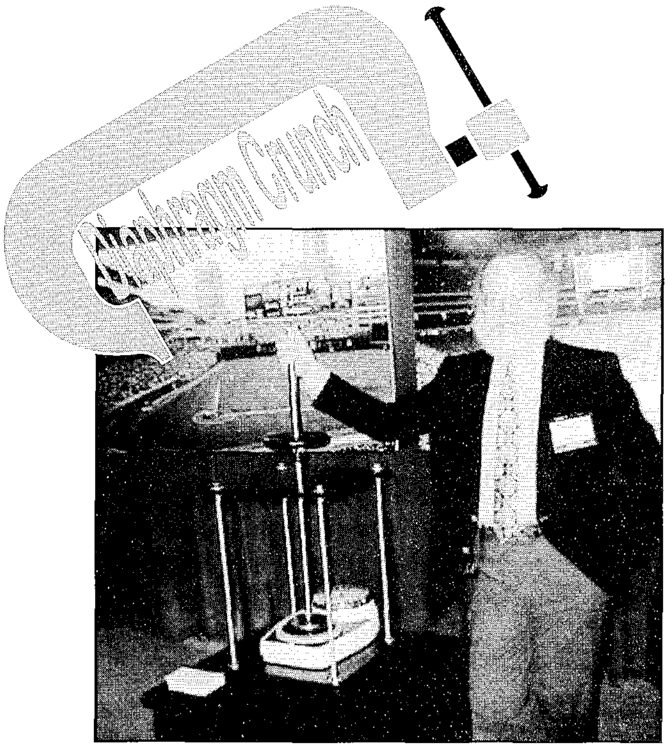
Figure 5b - EFC (energy-frequency curve) of the first 160+ ms of energy arrivals at the test mic position, which includes the direct and early-reflected sound fields.



The lower trace represents the direct sound field, while the upper trace includes the early-reflected field. One-third octave smoothing has been used for clarifying the display. Note that the lower trace (L_D) is room-independent, while the upper trace will change with the presence of an audience, etc. Equalizing the system using the upper trace as a reference is often referred to as "room equalization" even though the room is not being changed.



A comparison of the system/room response with and without the effect of the reverberant sound field. Note that the curves are similar at the high-frequencies, but there is some energy accumulation at mid and low frequencies. Both displays show why less amplifier power is needed indoors, since the recycling of energy from the room amounts to 6 -10 dB of "room gain." The listener hears the combined response of the direct, early-reflected, and reverberant sound fields. Equalizing the direct sound field is a science, while compensating for the effects of the others is considered by many an art.



What? Another new kids cereal? Actually it was a way to draw attention to the fact that something good just got better.

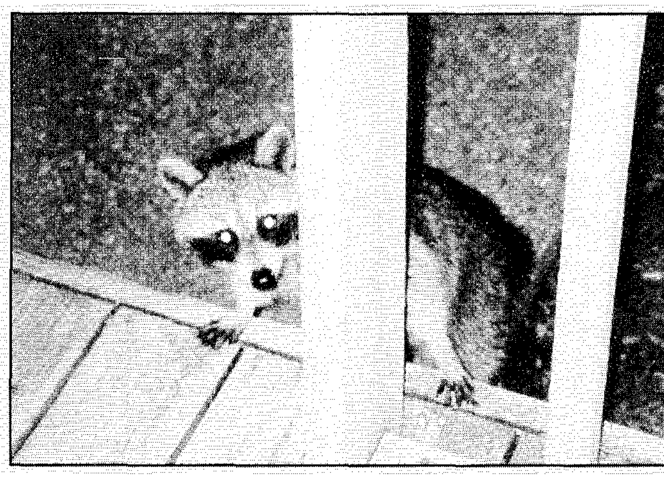
Community used the NSCA Show to formally announce their new diaphragm for the M4 driver. Already one of the most robust drivers on the market, the M4's aluminum diaphragm has been replaced with one made of carbon graphite. A contest was initiated to see who could guess the "crush weight" of the new assembly, one of which was officially smashed at 5 pm on Tuesday. Pictured at left is company president and chief designer Bruce Howze, along with the mechanism used to do the deed to the diaphragm.

The diaphragm collapsed (it didn't crush - when the weight was removed it bounced back to its previous shape) at 280 lbs. The correct guess was put in by Perry Gibson of Our Master's Voice Audio in Beaver Falls, PA.

It remains to be seen whether the upgrading of the M4 diaphragm will create a "vintage diaphragm" market among audiophile purists who prefer the sound of aluminum. (Ha!)

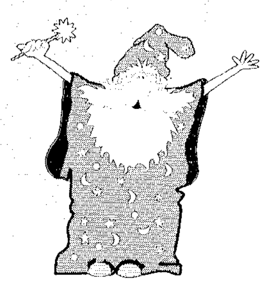
Nocturnal Visitor

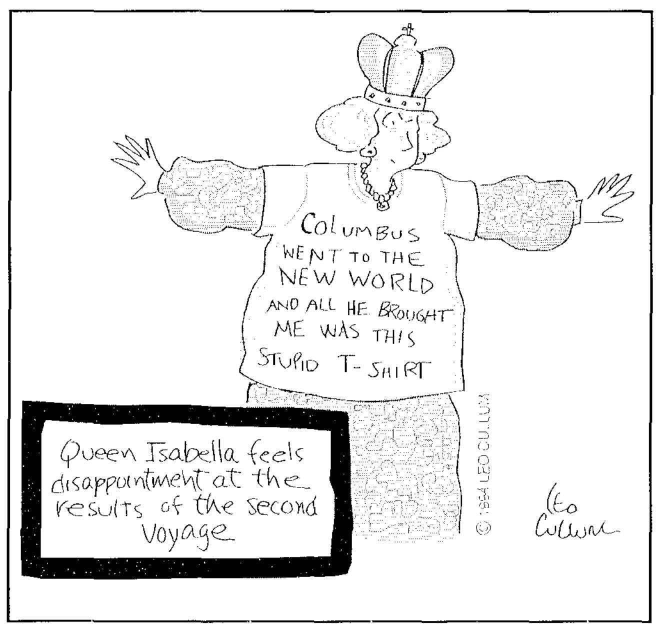
Racoons are plentiful in Southern Indiana, but rarely do they venture so closely to the house. This one came right up to the backdoor. Recent years have brought a tremendous increase in wildlife to the area. Deer and fox, rarely seen in past years, are common sightings. The girls recently ran across a 5-foot black rat snake on the driveway, which we captured and transplanted to a wooded area several miles away, which according to Brenda wasn't nearly far enough!



The Old Philosopher Says...

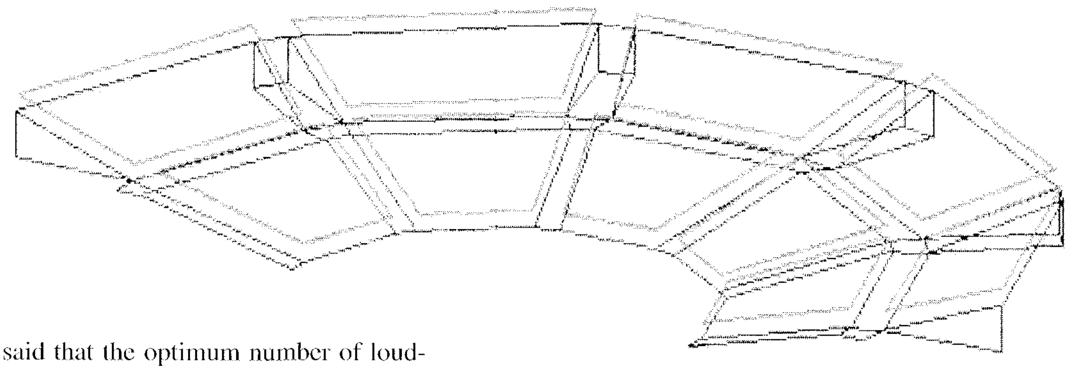
You can't make everyone happy, but you <u>can</u> make everyone mad!





An EASE Application

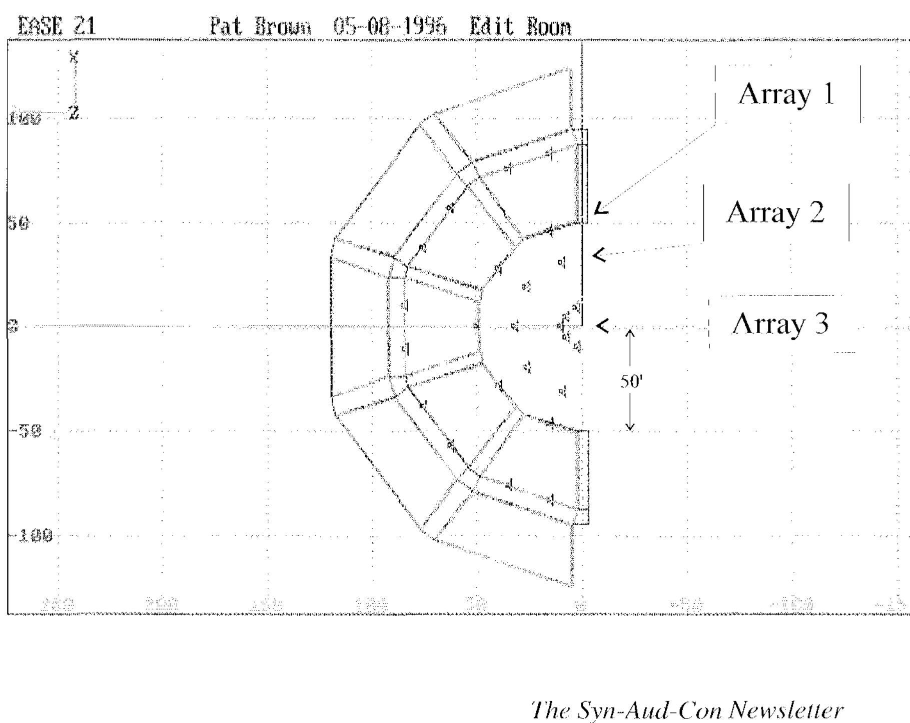
Working With Arrays



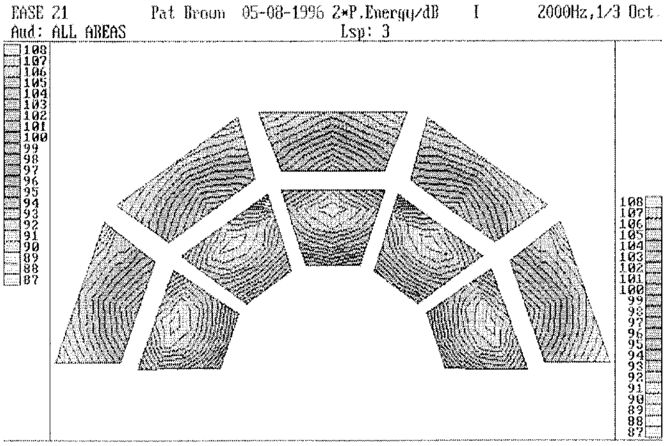
We have always said that the optimum number of loudspeakers is one, since multiple loudspeakers cannot occupy the same physical space, and as a result will acoustically interact with each other. The system designer's goal is not to eliminate the interaction (which would be impossible with present day devices), but to minimize it's effect on the sound delivered to the audience, or intentionally use it as a tool. Syn-Aud-Con seminars repeatedly demonstrate some methods that can help. One method is to minimize the overlap as much as possible, which involves the use of precision signal delays to synchronize devices at their point of physical overlap. Another method involves deliberate miss-synchronization to achieve a dense comb filter pattern that the human ear-brain system seems to smooth out nicely. The dense comb filtering can be achieved with signal delay, or with multiple loudspeakers, as in the case of a high-density distributed overhead system, or a brute-force concert array.

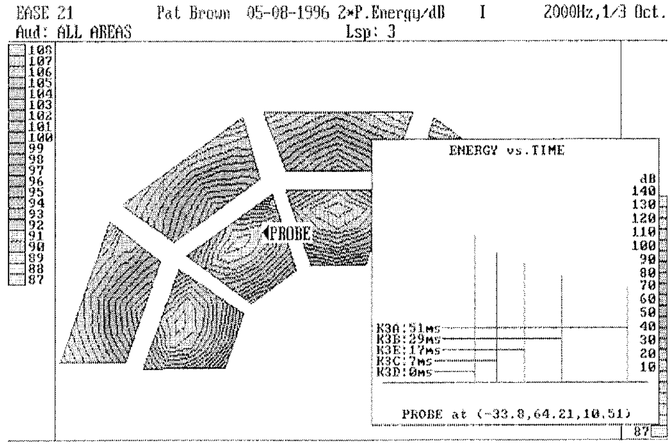
Ironically, the poorest performance is typically encountered in systems that fall between these two extremes; having multiple components but of insufficient density to smooth out the coverage. We will use the EASE program to demonstrate the problem and some possible solutions.

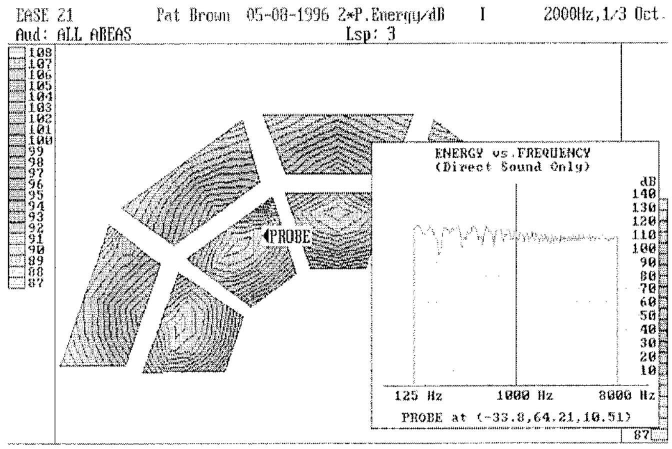
Consider an auditorium with audience areas as depicted above. The problem is how to deliver the best possible coverage to the five lower audience areas. While there are numerous possible approaches to the problem, we shall consider three possible ways to implement a 5-box array in the venue. Such systems offer many practical benefits, such as simplified rigging and compact size. Often overlooked are the acoustical ramifications of how such devices might interact with each other, which will ultimately affect the uniformity of audience coverage. The array configurations that will be considered are a single-point, tightly-packed array, an ex-



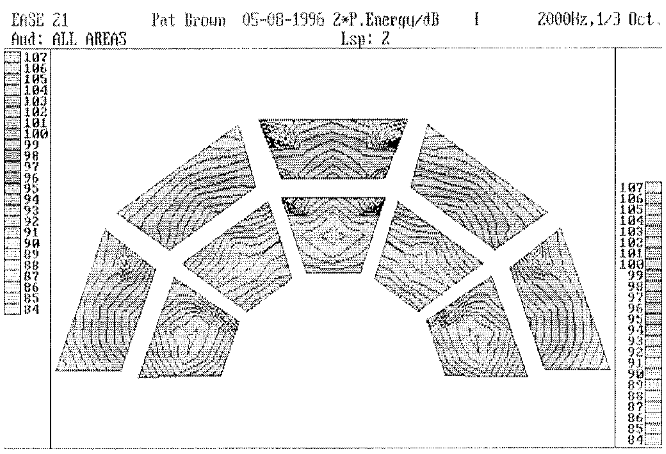
Plan view of auditorium, showing proposed array locations

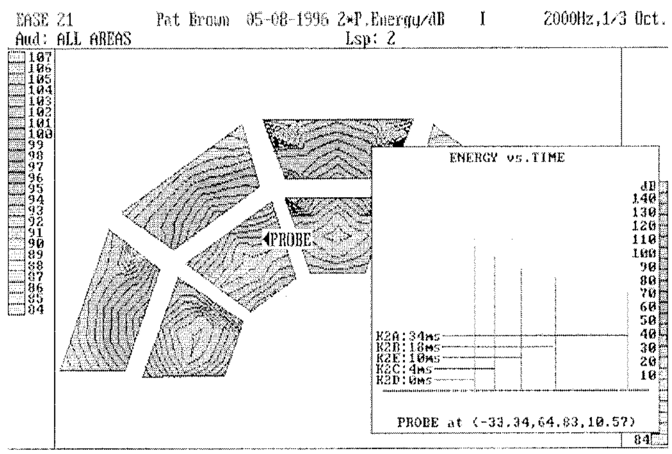


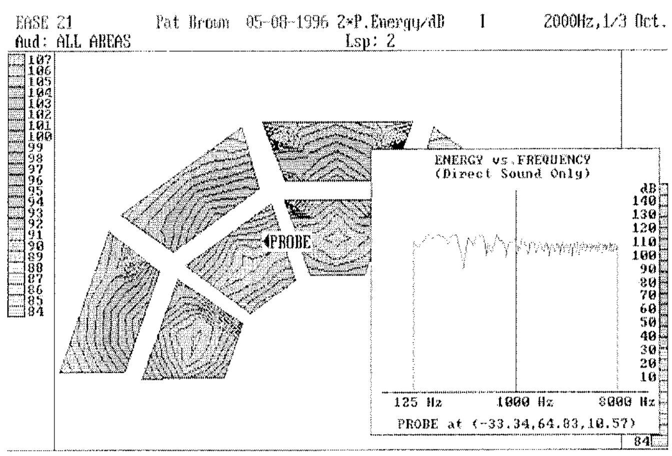




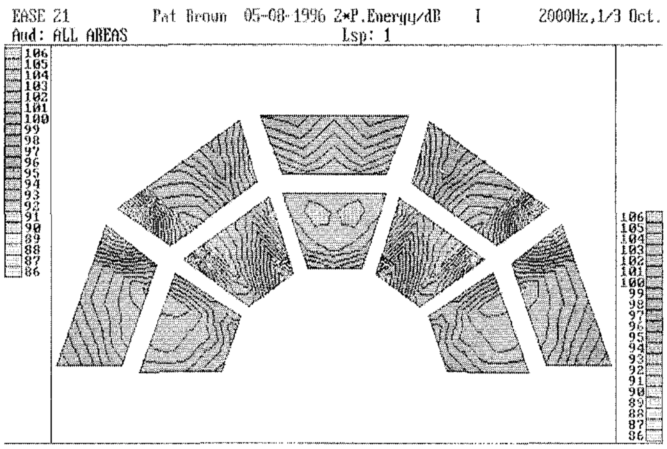
Model 1/ Array 1 - In this configuration, the loudspeakers are behaving independently of one another in the mid and high-frequency bands. Each seating area is receiving the bulk of its coverage from a single loudspeaker, with little interaction between devices, with the exception of some low-frequency coupling due to the long wavelengths at these frequencies. Since comb filtering is minimal, equalization should be a straightforward process. Note the time separation of the loudspeakers on the energy-time curve, and the resultant frequency response on the energy-frequency curve.

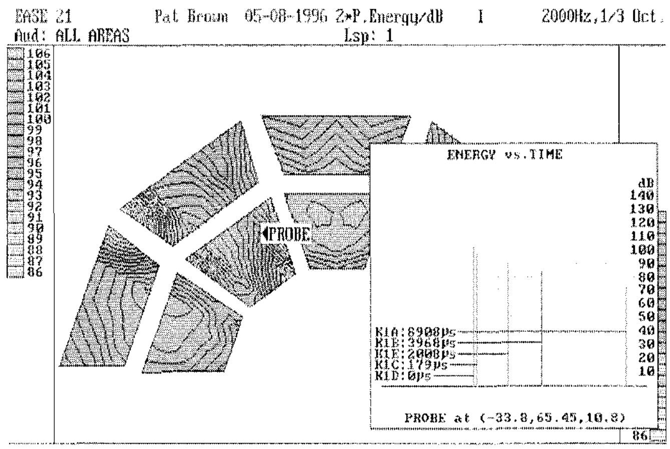


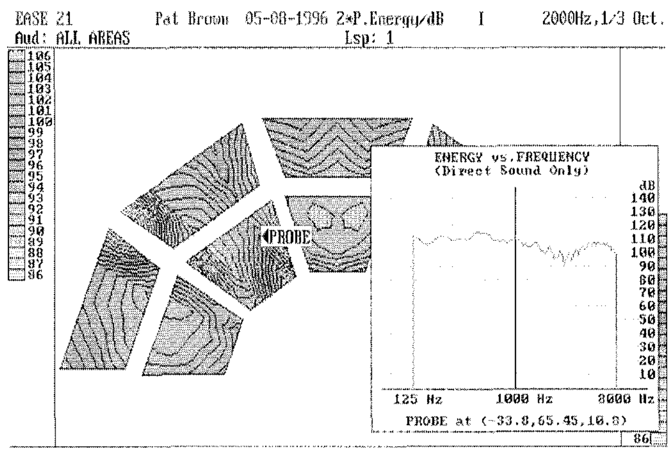




Model 2/ Array 2 - In this configuration, there is more interaction between the loudspeakers. Some listeners are receiving coverage from more than one loudspeaker, and the time offset is causing phase cancellation in those areas. Proper equalization of the system must be accomplished by turning off adjacent loudspeakers, since interference effects cannot be corrected with an electronic equalizer. Note the time separation of the loudspeakers on the energy-time curve, and the resultant frequency response on the energy-frequency curve.







<u>Model 3/ Array 3</u> - This configuration ensures the maximum interaction between the loudspeakers. All listeners are receiving coverage from more than one loudspeaker, and the time offset has produced severe cancellation in the vocal region. Low-frequency energy below the array will be excessive, possibly necessitating drastic equalization to achieve adequate gain-before-feedback. Note the time separation of the loudspeakers on the energy-time curve, and the resultant frequency response on the energy-frequency curve.

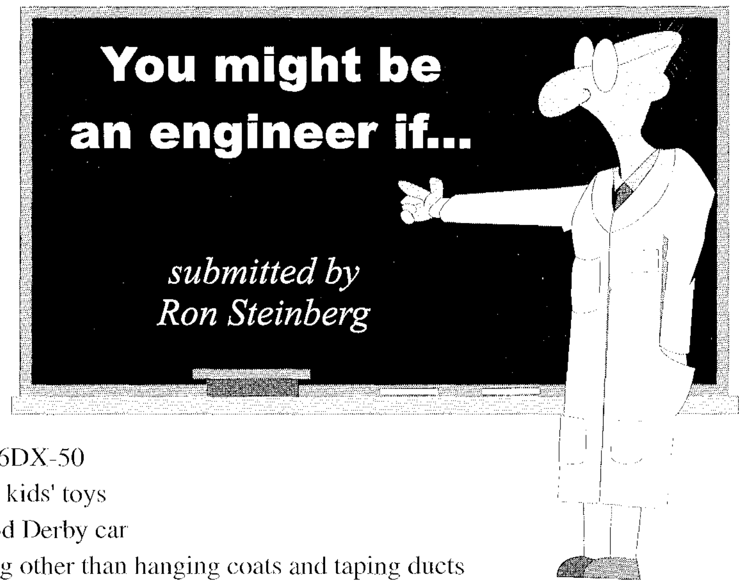
ploded version of the same, and a zoned-circumferential version of the same. The 2kHz octave band will be considered, because of it's obvious importance to speech intelligibility.

The first model shows the zoned-circumferential array. The loudspeakers are placed directly above the front row of each audience area, which represents the shortest practical loudspeaker/listener distance, as well as the maximum practical distance between devices. As would be expected, the interaction between devices is minimal. This is due to the fact that the physical location of each loudspeaker makes it the predominant source for that particular audience area. The energy arrivals from the other loudspeakers are lower in level (inverse-square law) and the comb filtering produced is tightly packed due to greater loudspeaker separation. While this may represent the smoothest coverage attainable, the down side is that it will probably be more expensive to implement, and may present some logistical problems such as interference with lighting and accessibility for servicing.

The next model represents the exploded array, which involves moving the loudspeakers about halfway back to a single point of geometric origin. Note that the well-defined coverage patterns of the previous case are showing signs of corruption, since the loudspeakers are now located closer together physically. Each area is still being covered by primarily one loudspeaker, but the energy arrivals from the other loudspeakers are higher in level and closer in time, resulting in broader and deeper comb filtering. All-in-all, this may represent a possible compromise between two extreme cases, retaining some of the advantages of both.

The last model is a prediction of what happens when the loudspeakers are located in a tightly-pack "banana" array, a common implementation in many auditoriums. Arrayed in this manner, the loudspeakers lose their individual acoustical identities, since their patterns now exhibit the highest possible degree of overlap. The coverage becomes random in nature, which will result in different sound at different seats. Since the interaction is frequency-dependent, the array will now have different coverage patterns at different octave-bands. A common problem with such arrays is the buildup of low-frequency energy under the array, which can make distant miking techniques difficult without drastic equalization. Benefits of this implementation include higher SPL (due to device coupling), simplicity of rigging, and a pleasing aesthetic look.

The role of the system designer is to analyze the needs of the client, and choose the implementation that best serves those needs. We are fortunate to have design tools such as EASE that allow advance warning of the problems that may be encountered with each of the possible configurations.



If what's in your shirt pocket weighs more than what's in your back pocket
If you introduce your wife as "mylady@home.wife"
If your spouse sends you an e-mail instead of calling you to dinner
If you can quote scenes from any Monty Python movie
If you want an 8X CDROM for Christmas
If you can name 6 Star Trek episodes
If your wrist watch has more computing power than a 486DX-50
If you look forward to Christmas only to put together the kids' toys
If you use a CAD package to design your son's Pine Wood Derby car
If you have used coat hangers and duct tape for something other than hanging coats and taping ducts
If you window shop at Radio Shack
If you've ever replayed a portion of a movie because you spotted a technical inaccuracy
If you have "Dilbert" comics displayed anywhere in your work area
If you are convinced you can build a phazer out of your garage door opener and your camera's flash attachment
If you know the direction the water swirls when you flush
If you own "Official Star Trek" anything
If you have ever taken the back off your TV just to see what's inside
If you own one or more white short-sleeve dress shirts
If you have never backed-up your hard drive
If you have ever saved the power cord from a broken appliance
If you see a good design and still have to change it
If the salespeople at Circuit City can't answer any of your questions
If you still own a slide rule and you know how to work it
If you own a set of itty-bitty screw drivers, but you don't remember where they are
If you rotate your screen savers more frequently than your automobile tires
If you have more toys than your kids
If you need a checklist to turn on the TV
If you have introduced your kids by the wrong name
If you have a habit of destroying things in order to see how they work
If the microphone or visual aids at a meeting don't work and you rush up to the front to fix it
If you can remember 7 computer passwords but not your anniversary
If you have ever owned a calculator with no equal key and know what RPN stands for
If you know how to take the cover off of your computer, and what size screw driver to use
If you can type 70 words a minute but can't read your own handwriting
If people hound you for pocket protectors at Halloween time
If your wristwatch has more buttons than a telephone
If you thought the real heroes of "Apollo 13" were the mission controllers
If you think your computer looks better without the cover
If your wife hasn't the foggiest idea what you do at work
If you spend more on your home computer than your car
If you know what http:/ stands for
If you've ever tried to repair a $5.00 radio

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

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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
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D. Wayne Lee, P.E.
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312-728-0565

OLSON SOUND DESIGN

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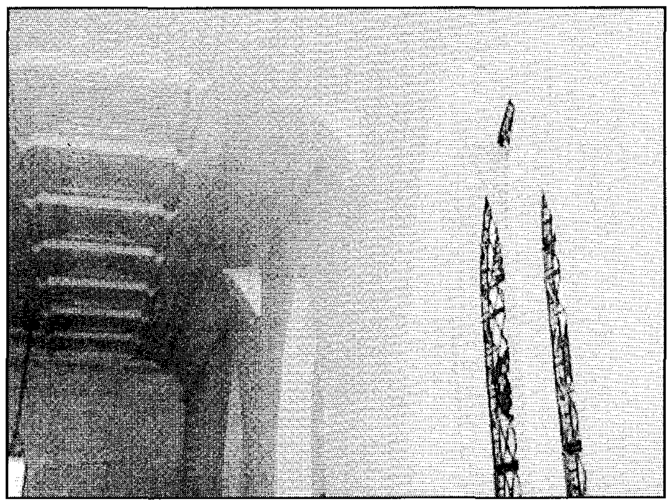
NOW THAT'S CUSTOM...

Where do you put the loudspeakers when the customer says that you can't touch the antique ornamental metal ceiling or obscure the stained glass windows? This case involved the custom fabrication of enclosures that were shaped like the windows. The distributed loudspeakers were combined with the Crown IQ System to provide sound reinforcement for a century-old small country church (about 100 members). This is a good example of how a small congregation can preserve the aestetics of their sanctuary, and have good sound without an operator present.



"Yes, we put on our pants the same way, Hayward, but mine are a lot more expensive."

"Congratulations, Fernbaugh. In just six months you've moved from an entry-level position to an exit-level position."

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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.

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Crown International, Inc.
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Eastern Acoustic Works, Inc.
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Buchanan, MI 49107
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616-695-1304 Fax

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IED - Innovative Electronic Designs, Inc.
9701 Taylorsville Rd.
Louisville, KY 40299
502-267-7436 Phone
502-267-9070 Fax

Manufacturer of computer-controlled sound system products and systems, including custom applications.

IRP, Inc.
1111 Tower Lane
Bensenville, IL 60106
708-860-1993 Phone
708-860-1997 Fax

Manufacturer automatic mixers, equalizers, power amplifiers, modular signal processing systems, transversal equalizers.

JBL UREI ELECTRONIC PRODUCTS	**JBL, Inc.** 8500 Balboa Blvd. Northridge, CA 91329 818-894-8850 Phone 818-830-1220 Fax	Manufacturer of horns, loudspeakers, drivers, enclosures, electronics, CADP2 system design program for Windows.
JWD	**J.W. Davis Company** 3030 Canton St. Dallas, TX 75371-0219 214-651-7341 Phone 800-388-9106 Fax	Manufacturer and distributor of loudspeaker products, electronics, microphones, installation aids, wire and cable, Pataxial loudspeakers, etc.
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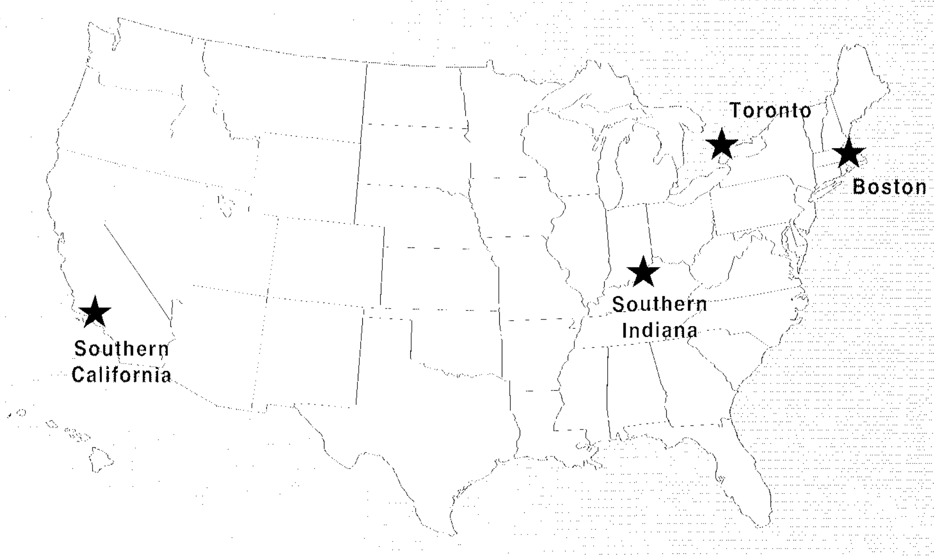

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