The Science of Audio - a series of lectures by Floyd E. Toole, Ph.D. Vice President Acoustical Engineering
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harman/kardon

Loudspeakers and Rooms for Multichannel Audio Reproduction

by
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Part 1 – How many loudspeakers? What kind? Where do we put them?

Here we look at the basic theory of multichannel audio systems, leading us to understand why certain loudspeaker designs and room arrangements work better than others.

67 years ago, in1934:

Bell Telephone Laboratories research concluded that there were only two ways to properly reproduce a realistic sense of direction and space:

1. Binaural – dummy head recordings reproduced through headphones. O.K. but not commercial.
2. ***Multiple channels –***

- How many? More are better!
- A realistic minimum? ***Three*** across the front to reproduce only the soundstage, not ambiance.

So, how many did we get?

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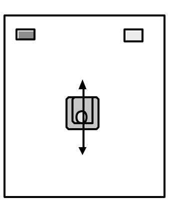
In the beginning, there was monophonic – single channel – sound. Compared to nothing, it was impressive, but eventually serious people began to wonder what more might be possible. Bell Labs, one of the world's greatest research institutions, looked into what was required for accurate directional reproduction, and concluded that, for loudspeaker reproduction, multiple channels were needed. While many channels were desirable, they were not very practical, certainly not at that time, so they looked into a practical minimum number. They came up with three front channels, just to reproduce the soundstage – no ambiance, no sense of spaciousness. For a single listener, they thought that two channels would be sufficient.

So, in the 1950's we got two channels.

Multi-channel Sound - 1st Try

TWO CHANNEL STEREO



***Better than mono**

***Antisocial - sweet "line"**

***Poor sense of ambiance or envelopment**

The reason was that there were no practical methods, at the time, to get more than two channels into and out of the groove of an LP record.

We have put up with this limitation for over 50 years, and it is time to move on to better things. The idea that we only have two ears, and therefore need only two channels, applies only to headphone listening. In normal hearing our two ears and brain give us a remarkable three-dimensional sense of direction and space. To reproduce this, we need many channels.

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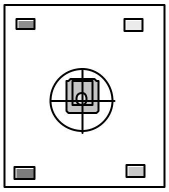
Multi-channel Sound - 2nd Try

QUADRAPHONICS



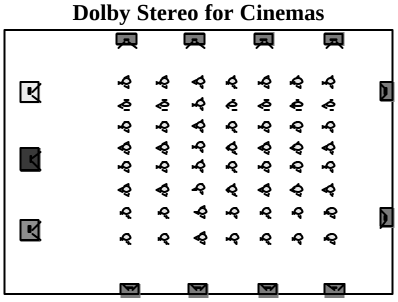
***More directional effects are possible**

***Better sense of ambiance or envelopment**

***REALLY Antisocial – Sweet SPOT!!**

In the seventies, there was an attempt to do better. It failed because of industry disagreements over which of several competing systems should become the standard. Frankly, I am glad it failed, because it was the wrong arrangement of channels. With no center channel, the annoying stereo sweet spot remained. Most of the systems had a lot of crosstalk, or leakage among the channels, so that even the front-back impressions depended on the listener being half way between the front and back speakers. With no side channels, the sense of ambience and spaciousness was less than it could be.
A lot of paraphernalia for another antisocial system.

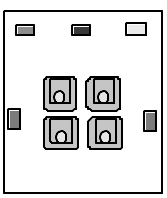
Dolby Stereo for Cinemas



The multichannel matrix technology underlying quadraphonics was quite clever, and Dolby seized on it for Cinema applications. They revamped the active matrix system into four new channels: a left, center and right front array, and a single surround channel that was sent to several speakers distributed down the sides and across the back of a cinema. They included features that allowed the system to function with the imperfections that pervaded optical sound tracks – then the basis of the film industry. One of the compromises was a severe limitation in the high frequency response of the surround channels.

Multi-channel Sound - 3rd Try

Dolby ProLogic - for homes



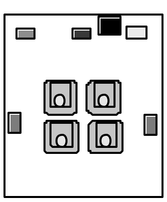
*Social ! It can be shared.
*A *real* center image

*Only fair ambiance and envelopment (mono surround)
*Surrounds not broadband
*Directional information only across the front

Naturally, the system migrated to the home, with the simplification that the surround channel was sent to a pair of speakers located at the sides of the listening area.
One of the most common mistakes is to locate the surround speakers behind the listeners, at the back of the room. The perception of spaciousness, or surrounding ambiance, is greatest when the sounds at the two ears are uncorrelated (different from each other). This is most effectively achieved when sounds arrive from the sides. This puts you in the hall, not between the band and the hall.

Multichannel Sound - 3rd Try

Dolby ProLogic - for homes



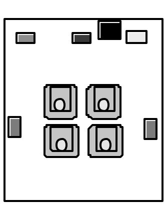
And the common-channel subwoofer arrives on the scene . . . and stays!

8

Recognizing that customers would react badly to being told that they needed five big full-range speakers, a subwoofer channel (the .1 in 5.1) was added.
However, in the beginning, no provision was made for a proper crossover between the subwoofer and the five "satellites". As a result, there were many unhappy experiences in trying to make a good sounding acoustical transition between satellites and subwoofers that were not necessarily designed for each other. Because of this, many people thought that subwoofers were inherently less good than full-range systems. They were often right.

Embellishments to ProLogic

Home THX



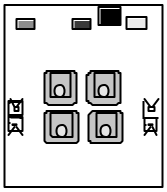
*low-pass and high-pass filters for sub and satellites.
*decorrelation of surround channel speakers
*"timbre matching" of surround speakers – a dubious feature!
*re-equalization (treble attenuation) of sound tracks.

9

Recognizing that cinemas were not always doing justice to their ambitious sound tracks, Lucasfilm set out to standardize sound quality in cinemas with their THX program. Then, sensing that things were not always right in the consumer world, they created a Home THX program, licensing a number of features and functions to manufacturers of surround processors. The loudspeaker crossover feature was welcomed, as was the electronic decorrelation of the surround channel, making the single channel sound less "monophonic". The timbre matching of surrounds was inappropriate, and the re-equalization was not always what was needed, but these functions could usually be switched off in the processors.

Embellishments to ProLogic

Home THX



*vertical directivity control of loudspeakers - dubious

*"dipole" surround speakers -

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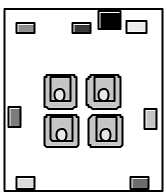
To add differentiation in loudspeakers, THX promoted the idea of restricted vertical directivity. In my opinion, if one wishes to restrict directivity, it should be done in both vertical and horizontal directions. Directional control is very difficult to achieve with cone and dome drivers, and still retain high sound quality, but it is easily done with horns and waveguides. Some basic performance standards were also a part of the THX program.

To help two monophonic loudspeakers to sound like more, the idea of "dipole"(bi-directional out-of-phase) surrounds was introduced. It helps in many situations. And, so do other multidirectional designs that spray sound in many directions.

Beyond 5.1:

Logic-7 (Lexicon, Harman/Kardon), Dolby ProLogic II



*FIVE or SEVEN Steered channels
*Directional information in all channels
*Superb ambience and envelopment

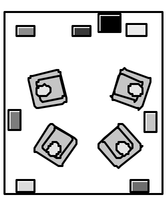
*Great for MOVIES & MUSIC

As time passed, the limitations of four-channel ProLogic became more evident. It was, after all, designed when optical sound tracks in films were the norm, and several features catered to problems with that signal source. A very successful effort to improve reproduction from film sound tracks was Logic 7, a sophisticated digital active matrix system that translated two channels into five or seven. In addition to all-channel steering, all channels were full bandwidth. Designed by Lexicon with a special sensitivity to the needs of music, Logic 7 has modes that tastefully extract and reproduce ambience from stereo recordings. Some of the modes are extensively user adjustable to accommodate personal tastes or system peculiarities. In my personal experience, the majority of stereo recordings benefit from a well-done multichannel conversion.

Recently, Dolby has introduced an upgraded ProLogic II with some of these features.

Beyond 5.1:

Logic-7 (Lexicon, Harman/Kardon), Dolby ProLogic II



*Sounds good over most of the room. You can look out the window!

12

The traditional 5.1 channel arrangement evolved from movie theaters, in which everyone is regimented into rows, facing the screen, in the dark. While there are occasions to do this in homes, the reality is that most of our entertainment does not require such restrictions. For more casual video entertainment and for music, we may have some light in the room, open the blinds to see the garden, and face away from the screen. With only two loudspeakers at the sides, the surround illusions suffer and one may be aware of the limited number of channels. With four surround channels, appropriately processed, the sense of space can be remarkably seamless for listeners seated over much of the listening area. And, because there are now four speakers, there is less of a need for them to be multidirectional. In very small rooms, multidirectional speakers may still be less easy to localize, but in larger rooms that problem disappears and we can make all channels absolutely equal in sound quality, something that is important with multichannel music recordings.

2 channels in / 5 or 7 channels out

- INPUTS: any stereo source
  - CD's, tapes, LP's
  - Television, VCR, Laserdisc, DVD
  - Stereo radio or satellite/cable audio channels
  - Games
- OUTPUT:
  - 5.1 or 7.1 multichannel audio

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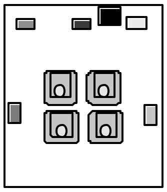
The entertainment business does itself a disservice by over emphasizing the "home theater" theme. We all watch movies, to be sure, but we spend far more time watching TV, listening to FM radio, CD's, satellite/cable music channels, etc.

We need to acknowledge the idea that a multichannel audio system brings a higher level of enjoyment from ALL sources of sound. A high quality two-channel to multichannel conversion system makes this possible.

The often gimmicky and mostly bad sounding "hall", "stadium" and "club" reverberation devices are not what I am talking about. These can be fun, but a tasteful multichannel conversion relies on ambience extraction, digging out the spatial information in the original recording, not adding synthetic reverb that may not be appropriate.

Multichannel Sound - 4th Try

5.1 channel Digital Discrete



Dolby AC-3
MPEG2
DTS
SACD
Audio DVD
etc.

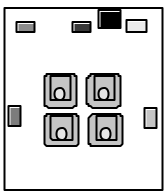
MOVIES

MUSIC

Eventually it had to happen, and it has. Technology caught up with our desires to have a fully discrete multichannel system, with all channels equal and independent. Now that it has, because it is digital, and therefore flexible, we have several systems. Not all of them need to survive, but at least all of them sound very good. SACD and DVD Audio, in certain of their modes with very high sampling rates and dynamic ranges, can be better than any human listener needs them to be. These excesses waste bandwidth, but it can certainly be said, with assurance, that the delivery medium is not a restriction to good sound. The recorded music itself is as variable as ever, so, don't shoot the messenger if you don't like the message.

Multichannel Sound - 4th Try

5.1 channel Digital Discrete



*Social - group listening
*A *real* center image for stability and better sound quality
*A good sense of ambiance and envelopment as required
*FIVE discrete channels
*Directional information in all channels
*All channels broadband and equal power

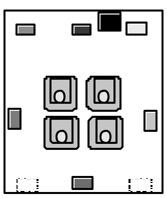
15

The industry is still learning. The medium is in good shape, it is the messages that are mixed.

Still, if five channels are good, more would be better.

Beyond 5.1 Digital Discrete

Dolby/THX EX, DTS ES, DTS 6.1 ES

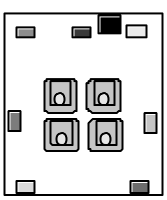


*A single center rear, or a pair of side rear, Loudspeakers for a matrix-derived 6th channel or, in DTS 6.1 ES, a discrete 6th channel.

Resurrecting a feature of ProLogic and using it to extract a center rear channel from the surround channels of a Digital Discrete 5.1 mix is a logical embellishment. The channel can be reproduced from a single center speaker or a pair at the rear sides.

Beyond 5.1 Digital Discrete

Logic-7 – Lexicon, Harman/Kardon

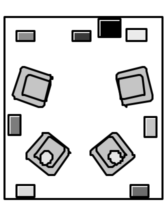


*SEVEN Steered channels
*Directional information in all channels
*All channels broadband
* Great for movies
* Great for MUSIC

As always, if one is good, two are better. Logic 7 retains compatibility with the derived center idea, but improves on the sense of surround by adding some separation between the rear channels.

Get a life!

Logic-7 – Lexicon, Harman/Kardon



Great for MUSIC
And, guess what?

You don't have to look at a blank screen!

18

Again, one can break away from the rigid cinema-seating format, and have a chance to visually share a dramatic or comedic moment with your fellow viewers. When listening to music, one can swivel the "lazy-person" chair around and look at the garden view, without sacrificing the sound illusions.

Always remember –

We are selling ***more*** than Home Theater.

We are selling ENTERTAINMENT

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Multichannel Audio Entertainment

- Movies
- Multichannel music recordings
- Stereo music converted to Multichannel
- Television converted to Multichannel
- Games

20

The challenge is to get people into situations where they can hear good demonstrations of multichannel entertainment, using good equipment. The entry-level in-wall/in-ceiling systems, and systems with inadequate tiny surround speakers can be impressive if one is easily impressed. However, those of us in the business need to show customers that much better sound is possible, and sometimes for only modest cost increases. Good in-wall speakers exist, but they are not inexpensive. Ceiling speakers are for background music systems in stores (I exaggerate, but only slightly). Ceiling speakers should never be used for the front channels, and used for surround channels only in homes that I am never invited to.

But, how were they recorded?

What are we trying to reproduce?

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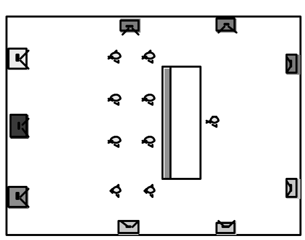
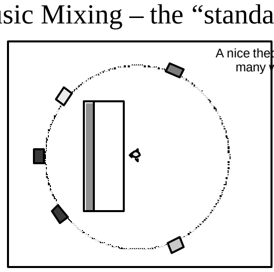
Film Sound Mixing/Dubbing Stage



22

Some folks make a big deal about replicating the cinema experience in the home. For me, I want better than that. First, I want quieter surroundings, better sound quality, my own volume control, the ability to pause the experience, a fridge and a bar for nibbles and drinks, and the ability to choose my company.
That said, the rest is easy - no big deal. Buy a popcorn machine. Film sound tracks are put together in a dubbing stage – a scaled down version of a cinema. The rooms are usually larger than we typically have at home, but otherwise there is nothing difficult about imitating the experience. Multiple or multidirectional surround speakers would be appropriate, and they would be located high on the walls.

Music Mixing – the "standard"



23

For music the rules change.

We are still in the early stages of multichannel music, and the industry is still figuring out what to do. One standard has surfaced, suggesting this arrangement of speakers. It is a reasonable approach.

All five speakers are intended to be identical. Some hard liners insist that all five be full range – not a good idea, as we will see later. They would all be at ear level.

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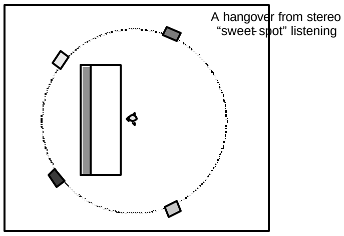
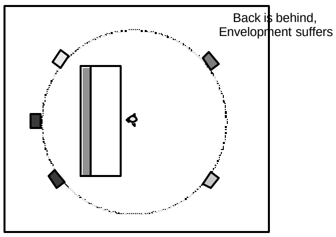
Music Mixing – "retro" surround



24

Old habits die hard, and entrenched stereo music engineers find it difficult to work with a center channel. So several of them simply don't use it, or use it very sparingly. The sweet spot is back.
They invent "reasons" like: the customer can turn off all of the other channels and then can hear "the talent" naked, without the backing. Another is that customers might buy small cheap center channel speakers, thus degrading "the talent".
Since 80% or more of what we hear in a movie is from the center channel, it seems to me that film people are the ones who should be concerned. In reality there is little to be concerned about. This is a regrettable practice that, I hope, will cease.

Music Mixing – Quad lives!



25

In looking for material to remix or "repurpose" into a multichannel format, it is natural that old master tapes from the quadraphonic era should find favor. Along with the music have come some old habits, namely placing the surround speakers towards the rear of the room, mirroring the fronts. If this is done for the engineering of the album, customers would have to move their surround speakers to the back of the room in order to hear what was mixed in their specific album. That is not going to happen, and this is a bad idea.

Repurpose the recording engineer!

The only agreement is about the LCR arrangement.

The Surrounds are up for grabs. No one solution will work perfectly for all programs!

26

THX introduced the idea of dipole speakers to surround systems, but there are many other ways to spray sound around a room. Actually, the THX speakers are not real dipoles, since the front and rear speakers are separated in space. Consequently they do not produce a sharp null in the direction of the audience, something that probably would be a disadvantage in any event.

Surround Speaker Options

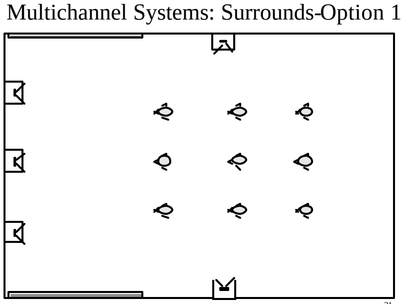
- Conventional forward-facing 'monopoles'
- 'dipoles'
- 'Bipoles'
- 'Tripoles'
- 'Quadrapoles'

  } Multidirectional radiators attempting to spray more sound away from the listeners than at them.
- Multiples of the above (more than one on each side wall, operating in parallel).
- Any of the above, used with additional derived and steered 'channels'.

30

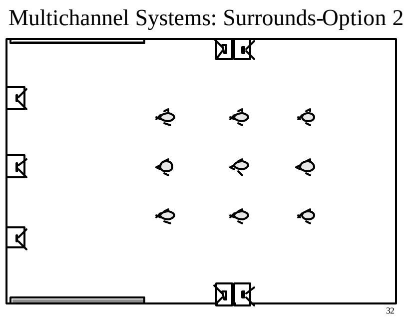
Since the THX "dipole" null is a soft one, it means that other multidirectional designs can achieve similar effects. Some of the JBL Synthesis dipoles go further and use horn loaded mid/high drivers to improve on the directional performance by reducing comb filtering in the null region.
With digital discrete and the new generation of active matrix systems, multidirectional speakers are less advantageous, and with seven channel systems, the need is even less.
Music oriented systems should use similar or identical speakers in all channels.

Multichannel Systems: Surrounds-Option 1

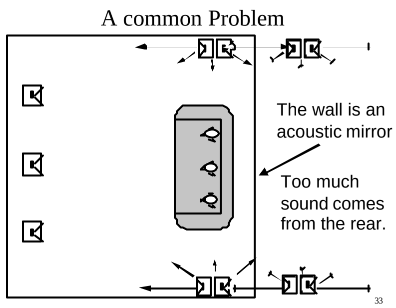


31

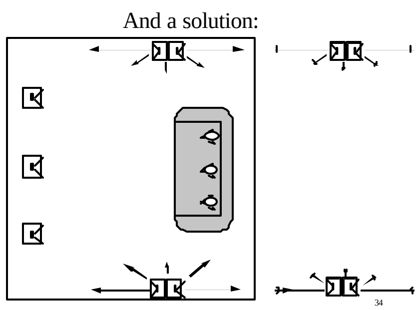
Conventional forward-facing loudspeaker systems work well in large rooms but, in small rooms, it may be too easy to localize them. Placing them high on the walls was an answer in the days of Dolby ProLogic, when surrounding sounds were mostly ambient and reverberant sounds and aircraft flyovers. Nowadays, they may need to come down closer to ear level for those occasions when we play multichannel music.
In-wall speakers? These can work, if you choose ones that sound really good (not cheap). Ceiling speakers? Not for me! Be careful, in-wall speakers can also entertain the room next door. Back boxes (enclosures for in-wall speakers) can help.



For small rooms, or when the surround speakers need to be close to the listeners, multidirectional speakers can be useful.
However, there is a problem that is not much discussed. Most of the sound in the surround channel in Dolby ProLogic encoded films is intended to be ambiguously localized – to come from everywhere and anywhere. Atmospheric music and reverberation were common. However, nowadays, all kinds of sounds are sent to the surround channels. Impulsive sounds, like gunshots and ricochets, are very unlike "atmospheric" sounds. They are usually intended to appear to come from certain directions, and they can be especially well localized by our keen ears and brain.

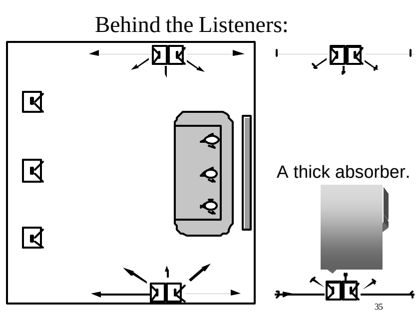


It is annoying when these sounds appear to come from the front (reflected from the screen or front wall) or rear, when they were intended to be at the side.
Lots of homes force us to sit with our backs to the wall. This is a problem, because the wall behind the head is an acoustic mirror. This setup often leads to sounds that were intended to be ambiguously localized, or deliberately to one side or the other, being localized behind the listeners. I have heard this in installations where they should have known better.

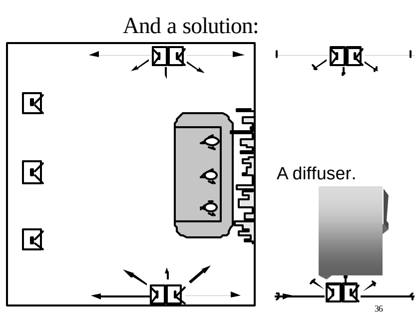


Moving the speakers forward pushes the mirrored speakers further back, improving the situation. Then be careful about reflections from the front surfaces.

It is sometimes suggested that the solution is to absorb the front and back wall reflections from the dipole surround speakers. Doing this leaves the listener in the direct sound field from the side speakers, which is just what we were trying to avoid by using dipoles. Incidentally, the sound quality to the sides of dipole surround speakers may not be as good as it could be.



Let's focus on the front speakers for a minute. A reflective wall behind the listener is a huge problem, because very shortly after the direct sound from the front arrives at the ears, there is a powerful reflection from the wall. Comb filtering degrades the s ound quality, and the temporal confusion degrades the soundstage imaging.
An absorber, like cushions, fiberglass, an acoustic foam panel, or the like, behind the heads brings the front soundstage into sharper focus. How thick an absorber? Not less than 2 inches. Thicker is better.
This is something that really works for stereo, where most of the soundstage is created from "phantom" images. Try it.



A diffuser behind the heads confuses the imaging even more. Not a good idea. It is well known from scientific research that uncorrelated sound at the ears (what the diffuser creates) delocalizes (makes less clear) auditory images in the front soundstage.

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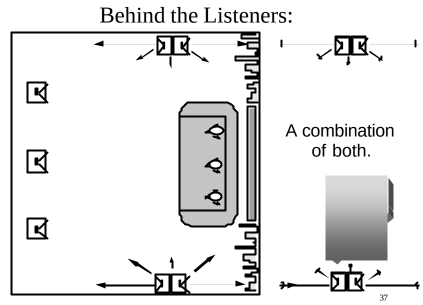
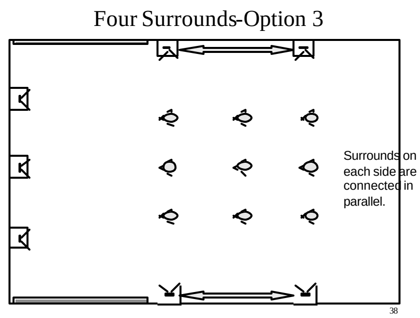
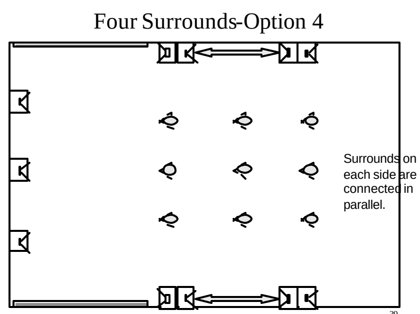


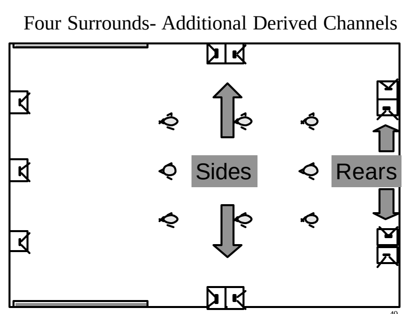
A bit of both can work though, with the absorber maintaining the integrity of the front soundstage, and the diffusers at the sides contributing to an enhanced sense of space.



In a large room, multiple speakers can be useful. Just hook them in parallel, the way they do in cinemas.

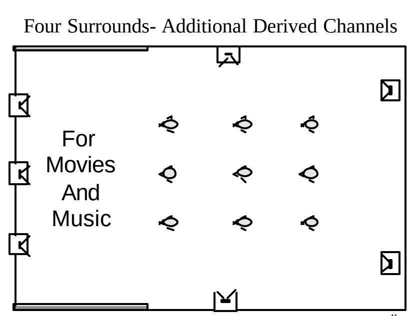


Multidirectional speakers can also be used, but watch out for sounds appearing to come from unusual directions because of reflections.



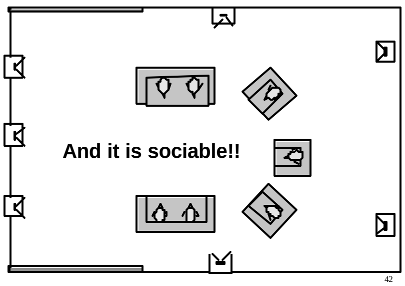
Still better, use a surround processor that creates two new channels with the appropriate delays and spectral processing. The two additional speakers now have their own signals, and the additional benefits are profound.

It works with multidirectional speakers . . .



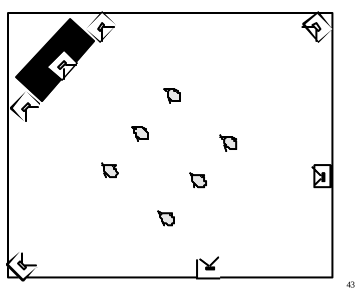
. . . and arguably even better with conventional speakers. In a largish room, this is a superb solution. If the side speakers are very close to listeners, one might use multidirectional speakers on the sides only.

Four Surrounds- Additional Derived Channels



The big bonus, from my perspective, is the liberation of listeners from the rigid theater-seating plan.

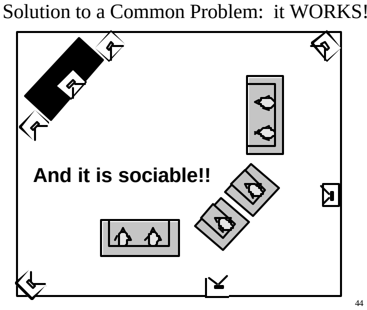
A Solution to a Common Problem: it WORKS!



It amazes me that so few people think to do this. If you are not building a special entertainment room from scratch, the probability is very high that there will be a fireplace, a door, a window, or an opening, precisely where the video display should go. It has happened in my last two houses, in spite of my attempts to avoid it.

So, just go with the flow, and put the system on the room diagonal. Those bothersome side-wall reflections go away, the side channel loudspeakers are a long way from the listeners – if they are in the corners, equalization will be necessary, otherwise move them down the walls a couple of feet or more.

A Solution to a Common Problem: it WORKS!



Having lived with this, I have come to really like it. The space behind the TV can hide a subwoofer, or a rack of equipment, or both. It is excellent for a general-purpose entertainment room. Now, I would not hesitate to deliberately design a room in this configuration.

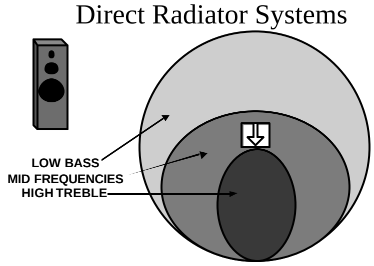
What is the ideal loudspeaker directivity?

Let's look at the options.

Since choosing a surround speaker type and configuration is difficult, one would hope that the choice for the left, center and right fronts should be much easier. It is.

So, what kinds of loudspeakers are best for the purpose? In the stores there are several options vying for our attention, and some are dramatically different in design from others.

Directivity of Forward-Facing Direct Radiator Systems



The traditional form of loudspeaker is the forward-firing direct radiator configuration. Woofers handle the low frequencies, tweeters the highs and, in some designs, a midrange fills in the intermediate frequencies. All drivers are arranged on the front face of the enclosure, facing the listeners. Because of their long wavelengths, low frequencies radiate equally well in all directions. As the wavelengths get shorter, in the middle frequencies, the radiated sound begins to favor the forward direction. At the very highest frequencies, the short wavelengths tend to radiate rather specifically in the forward direction. This is by far the most popular of all speaker design configurations.

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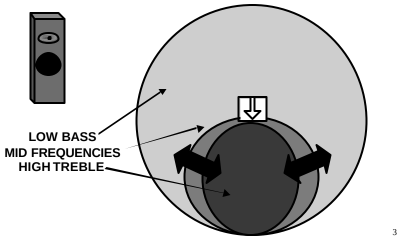
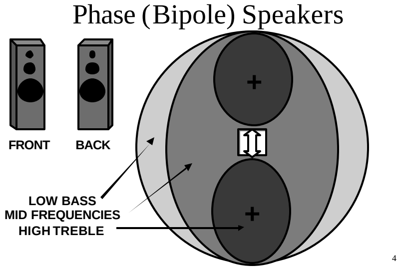
Directivity of Forward-Facing Systems with Horns or Waveguides



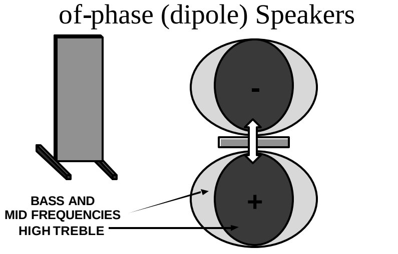
We now know that it is important that speakers exhibit uniform directivity through the middle and high frequencies. Consequently, one increasingly sees systems with horns or waveguides. A true horn, driven by a compression driver, is derived from high power professional systems. Waveguides (really they are horns) on tweeters are becoming more common as designers learn the advantages they offer. No longer need we concern ourselves with the old idea that horns have a “megaphone” kind of sound. Good horn designs do nothing but allow us to control the dispersion of the sound, making it wide or narrow, as we wish, in order to arrive at a superior system design. Designed properly, a horn can improve a system design.

Directivity of Bidirectional In-Phase (Bipole) Speakers



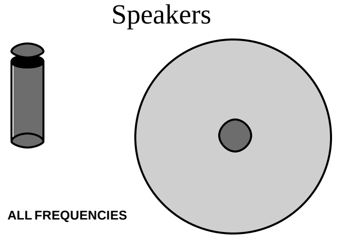
Putting two conventional systems back to back in the same enclosure, and letting them radiate in phase with each other, creates a system that is almost omnidirectional at low and middle frequencies. It generates a lot of reflected sound in a listening room, which, for stereo, is often a very good thing.

Directivity of Bidirectional out-of-phase (dipole) Speakers



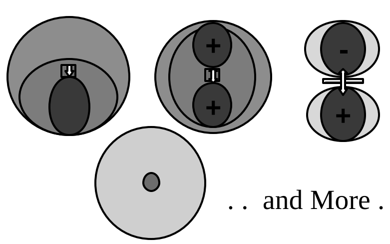
Flat panel loudspeakers of all kinds, electrostatic or electromagnetic, and some systems that use cones and domes, radiate sound equally in both directions, but do so in opposite polarity. This means that, where the sounds from front and rear meet, at the sides, they cancel each other out, creating a null.

Directivity of Omnidirectional Speakers



And finally, there are systems deliberately designed to be as truly omnidirectional as possible.

All of these . . .



. . and More . . .

. . . evolved during the era of two-channel stereo.

- They sound different from each other.
- Reflected sound from the multidirectional designs enhanced the sense of air, space and depth from stereo recordings.
- What suits one style of recording, may not suit another.
- Only forward-facing radiators are used by the artists and engineers creating the recordings.

8

Those of us who grew up with stereo, remember the numerous devices and techniques that were developed in attempts to create a more convincing illusion of space and envelopment. At the other end of the chain, in the studios, recording engineers experimented with microphone techniques and electronic processing to try to achieve the same end. All of this was trial and error. Among the variables that a consumer could play with was loudspeaker directivity. By spraying sound in more directions it was possible to create a greater sense of space, width and depth. Sometimes, with some recordings, it may be a bit too much. But that is stereo, always a bit of a gamble whether the recording technique and the playback technique match.

Multichannel Audio Changes the Rules

- With multichannel sound, impressions of direction, space and envelopment are in the recordings & sound tracks. The artists have more control.
- Customers at home have a higher probability of hearing what was created.
- *Some loudspeakers, designed to embellish stereo, are not appropriate for multichannel audio.*

9

Multichannel audio promises more. The existence of more channels is a big step in the right direction, but there are still opportunities to confuse the issue. With more discrete channels, the listening experience should translate from the studio to the home with much less distortion. The effects of the listening room should be less than in stereo, assuming that the loudspeakers don't unnecessarily aggravate the situation. In cinemas listeners are placed in a strong 'direct' sound field through the use of directional horn speakers up front, and acoustically damped rooms. In homes this argues for forward firing, if not horn/waveguide-loaded, speakers for the Left/Center/Right locations.

Speaker Mythology

Aren't some speakers better for movies than they are for music?

THIS IS THE WRONG QUESTION.
The real issue is which speakers are appropriate for stereo and which for multichannel reproduction.
The same multichannel systems can – indeed they MUST - work superbly for both movies and music.

10

In music recording, it is assumed that all monitoring will be done with conventional forward firing systems in all five locations.
But what of the persistent assertion that some speakers are better for movies than they are for music? The implication often is that we can get away with less 'refined' sound in movies than we can in music. But . . . there is music in movies – sometimes a lot of it. Sometimes, as in a concert video, the music is the entire point of the production. The assertion is silly. Good sound is good sound, whether it is in movies or music-only performances. The only special considerations for film sound are maximum loudness and power handling. In films, things occasionally can get very loud, especially in the low bass.

And the answer is:

- Forward-facing radiators
  - The overwhelmingly popular choice for stereo reproduction.
  - The only choice for monitor loudspeakers in making music and film sound recordings, both stereo and multichannel.
  - The logical choice for L,C & R in multichannel music and movies, and for surrounds in music.

11

What about directivity?
Multidirectional speaker designs tend to alter the spatial aspects of what we hear by adding reflections to the sound field. Is this a good thing for multichannel audio?
The evidence suggests that using anything other than forward facing speakers is a deviation from the principle of recreating the recorded "art" as it was intended.
My vote goes for conventional forward-facing systems.

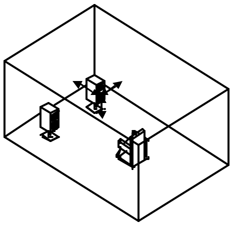
Locating Loudspeakers – some details that can matter a lot.

- Adjacent boundary interactions
- Speakers in bookcases and A/V furniture
- On-wall designs
- In-wall, in-ceiling mounting

1

So, now we know what kind of loudspeaker to use in each channel, and we know, more or less, where to put them. But there is more. The details of placement also matter if truly good sound is desired. Loudspeakers acoustically interact with the floor, and walls that are nearby. Small loudspeakers can be put on shelves with their backs against the wall, or even inside cavities in a cabinet. Doing this changes how they sound, sometimes seriously.
There are also on-wall or in-wall speakers that require no visible support, and are visually less obvious. Are these good choices?

Adjacent Boundaries: Optimizing Locations



Three dimensions are relevant. Each one is measured from the center of the woofer to one of the adjacent boundaries – floor and walls.

Each of these dimensions is related to a dip in the frequency response of the sound we hear in the room.

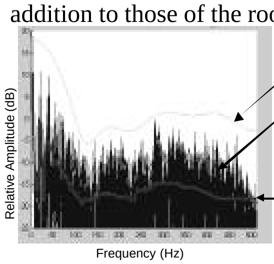
To minimize the effect on what we hear, try to make all three dimensions different from each other.

The general exception: woofers and subwoofers operating below about 80Hz. More about this later.

2

The location of a full-range or satellite speaker with respect to the adjacent boundaries determines the acoustic "loading" of the woofer. The consequence for listeners is that, at frequencies determined by the distances from the woofer to the reflecting surfaces, there will be variations in the sound radiated by the woofer. This usually shows up as peaks and dips in the frequency response – certain sounds being more or less loud than they should be. These effects cannot be eliminated, but they can be considerably reduced by making the distances to each of the nearby surfaces different from one another. The worst situation is if they are all equal. We will deal with subwoofer locations later in this paper.

The adjacent boundary effects are in addition to those of the room resonances.



For a specific speaker location:

The calculated effect of the adjacent boundaries.

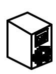
The superimposition of many 'room curves' measured within the listening area. These include resonances.

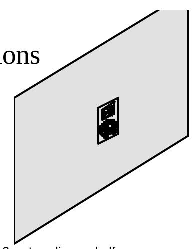
The average of all 'room curves', showing basic similarity to the shape of the boundary effect curve.

3

In listening rooms there are also resonances or standing waves. We will spend a lot of time on these later in this paper. However, for now it is important to realize that this is an additional effect. In this diagram can be seen a calculated curve showing the ups and downs in frequency response contributed by the adjacent boundaries. Under this can be seen the 'grassy' looking dark area which is the combination of many measurements made at different locations in the listening area of the room. The spikey 'grass' is evidence of variations caused by the room resonances – they are very large. An average of all these curves has been calculated and it can be seen to follow the general shape of the predicted curve. It is a real effect.

4p and 2p conditions





4p steradians = full free space
No reflecting surfaces nearby.

Most freestanding speakers are designed with this in mind.

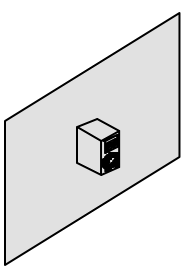
2p steradians = half space e.g. speaker mounted in a wall.
Low bass will be boosted compared to 4p conditions, and this is included in the design of in-wall speakers.

4

When speakers are designed, it has been traditional to evaluate them in what we call the 'free field', a space without any reflections. Technically, this is called a 4π environment, one where the sound from the speaker is free to radiate in all directions. We know this is not how they get used, but it is important to understand how the speaker functions alone, before adding the complications of the room. If a speaker is flush mounted into a wall, it is said to operate in a 'half space', or a 2π environment. This means that all of the sound that would have radiated to the rear is reflected back towards the front. If you look back a page or so, to the directivity of forward-facing speakers, you will find that the bass is omnidirectional.

The middle and high frequencies have a more forward directional bias. Consequently, mounting such a speaker in a wall causes the bass to be increased, something that must be compensated for in the design of speakers used for this purpose.

An Adjacent Wall



A conventional free-standing speaker used in this manner will exhibit increased low bass and colored upper bass. The speaker has been "redesigned", and needs equalization.

However, speakers designed specifically for on-wall use will be right at home.

For example: Infinity OWS1 (On Wall Speaker)

5

If a conventional box speaker is placed adjacent to a wall, the bass will also be increased and, depending on whether this was the intended location, the sound may or may not be correctly balanced. If the speaker is thick or deep, there may also be upper bass/lower midrange coloration due to interference of the direct sound with sound reflected from the wall behind.
Speakers designed for on-wall mounting avoid the problems entirely.

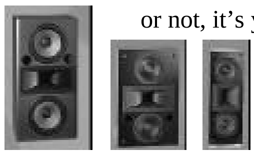
Freestanding Speakers get Built In

Were these properly measured and equalized after installation? I'll bet they weren't!

6

Custom cabinets often put speakers into specially made cavities. In the picture shown here, the speakers are, in effect, 2π mounted. The notes accompanying this photo tell us that the speakers were models that would normally be used in a free-standing manner. This installation substantially changes their performance. Low frequencies would increase in level, and since the cabinets no longer have any front edges or corners, the directional effects of diffraction are absent. To restore them to anything like their original sound quality requires equalization of a kind that is difficult to do in a room. This forces the installer into the role of speaker designer, something both he and the original speaker manufacturer might be very uncomfortable with!

Some speakers are designed to be built into custom cabinetry – or not, it's your choice.



Plain black boxes, with flat fronts and directional control to minimize diffraction effects. Final acoustical compensation for mounting differences are accommodated in the final installation step – equalization done by a trained installer using a <u>real</u> measuring system loaned by the manufacturer.

JBL SYNTHESIS

7

Speakers designed with large front baffles and some directional control are less susceptible to changes in performance due to their acoustical surroundings. Such speakers make life easier for custom installers, who have to adapt to the conditions of each customer's taste and physical circumstances. These 'black box' speakers are extremely versatile. The final step is to measure and to parametrically equalize the system after it is in place. In the JBL Synthesis systems this is done by a trained installer, using a computer-based multi-microphone measuring system that is loaned by the manufacturer. Knowing the 'target curve', how each model of speaker <u>should</u> measure is the key, and only the manufacturer knows.

Some "bookshelf" speakers actually work properly when placed on bookshelves . . .

. . . and have adjustments to allow them to work properly when on stands.

7

Small speakers are sometimes called 'bookshelf' speakers because they can fit into a bookshelf – albeit sometimes a rather large one. In actual fact, most such speakers are recommended for use on stands, positioned some distance from the walls.

However, if the speaker is powered, and if it has the appropriate equalization options built into it, the customer has the option of using it in either manner.

These got built in with a wood mesh grill in front. Two problems!

It looks O.K. but good speakers would be seriously corrupted!

8

Good woodworkers like to work wood. Here the craft has been taken too far, because of the coarse wooden grating over the speaker openings at the top of the cabinet. The open area is too small for this to be an acceptable grille.

The second problem is the elevation. Front channel speakers should not be so far above ear level.

Each speaker has its own little resonant cavity. B-a-a-a-d idea!

Manufacturers take care to avoid resonances in speaker drivers and enclosures. Here the empty space around the speakers in the cabinet add numerous unwanted colorations, honks and booms.
Done properly, the empty space would be filled with sound absorbing material and a grille cloth would cover the opening.

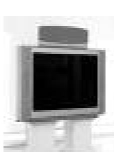
9

Speakers surrounded by empty space in an overly large cavity will have their sound degraded by the acoustical resonances in the open spaces. To avoid this, fill the empty space with fiberglass wool, or other fibrous material, foam, cloth or even old socks.

Cover the filling, or the entire opening with a porous, open weave, grille cloth. This is available in many colors at specialty shops or at any fabric store as polyester double knit.

Center-Channel Speakers

Can we use regular speakers?
It depends. Center channel speakers are (or should be) designed to sound good when placed on a television. The television modifies the performance of the speaker.



And some classy ones have adjustments to optimize their performance on different sizes of televisions!

10

Most center speakers find themselves sitting on top of a television set. Such speakers need to be designed with this in mind. Sadly, many are not.

The television set is an acoustical obstacle in the sound field of the speaker, and it modifies the sound of the speaker, mainly at lower frequencies.

Certain speakers can be adjusted to perform properly when used in a free-standing fashion, or sitting on TV screens of different sizes.

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In-wall and in-ceiling speakers

11

Speakers intended for in-wall or in-ceiling installation began as relatively inexpensive devices intended for distributed sound applications – music everywhere. As such, sound accuracy was not the first requirement. Many were direct descendants of car audio speakers.

Times have changed, and now we find installations in the main entertainment area of a home that are completely or partially outfitted with these speakers.

The virtues of in-wall speakers

- They are less visible

12

Out of sight, out of mind, the saying goes. Invisible speakers don't exist, but these come the closest, and as such fill a need for customers who just want the sound, but not the sight, of speakers. Let's put them in the ceiling. We rarely look up, right?

I accept ceiling speakers in airline terminals, and tolerate them in "music everywhere" uses in homes.
I do not endorse ceiling speakers in multichannel music or home theater systems.

The problems with in-wall speakers.

- Many of them don't sound all that good.
- Walls make poor speaker enclosures.
  - They resonate mechanically and acoustically
  - They can buzz and rattle
- Sound leaks through the wall and through ceiling/attic spaces into adjacent rooms.
- Many of them are located in the wrong places for good sound.

13

Mounting a speaker on a wall may be asking for trouble. Mounting one <u>in</u> a wall is worse. Of course the wall panels will vibrate, and radiate sounds that were never intended, corrupting the sound from the speaker itself. The vibration will occur on the opposite side of the wall as well, radiating sound into the adjacent room. A ceiling speaker will fill the attic with sound, letting it leak into all of the rooms on that floor.
The constraints of building frames and services often restrict where such speakers can go, and customers often ask for them to be put where they are least visible. Many end up in the wrong locations.

Hi-Fi in-walls + a difference

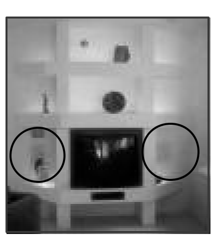
These compete with the sound of free-standing speakers, in part because of built-in mechanical isolation to reduce the wall vibrations generated by them.

15

Applying the same design criteria to in-walls as get applied to top line free-standing speakers, it is possible to make them into truly hi-fi products. Naturally, the problems of walls are still there, so it is a good idea to get inventive and find ways to mount the speakers so that the vibrations generated by them are not well coupled to the wall. This moves this category of product into the front line, able to compete with conventional speakers for multichannel sound applications. Place the front speakers close to ear level, and the surround speakers slightly higher. Please keep them out of the ceiling.

A good idea?

Only if the customer is not interested in high quality sound.



14

These poor in-walls are not just too low, but they are built into cavities. The additional touch of sculptures and ornaments that further obscure the sound truly tells us that sound quality was not a priority. Pity.

It looks good though.

Back boxes

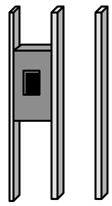
These are special enclosures designed for in-wall speakers create more predictable acoustical environments.

They also can reduce the amount of sound that leaks through the wall.

Do they guarantee good sound? No.

Are speakers without back boxes necessarily less good? No.

Back boxes are an option. Follow the manufacturer's recommendation.



15

The enclosure is an important part of a conventional free-standing speaker. It seems logical, therefore, to add an enclosure to an in-wall. There are difficulties, because the space is very limited in the space between the studs and wallboard. Adding a substantial enclosure made from thick materials, as is done conventionally, reduces the space further, and shallow enclosures are acoustically not a good thing, as a rule. Small enclosures are also problems if good bass output is required. So, some systems recommend back boxes, and others choose to live with the reality of walls, encouraging builders to add a few more screws and perhaps some glue, and filling the wall cavity with fibrous material. Both, demonstrably, can work.

Coming up in Part 2

What makes a loudspeaker good? Can the same loudspeaker sound good in different rooms? What about "acoustical" treatment of rooms? Can I customize a room to suit my personal tastes?

Loudspeakers and rooms operate as a system. One cannot be separated from the other. Knowing this allows us to design better loudspeakers, ones that are 'friendly' to different rooms. Knowing something about room acoustics allows us to use furnishings and acoustical devices to improve stereo and multichannel reproduction, making the listening experience more pleasurable.

harman / kardon

Loudspeakers and Rooms for Multichannel Audio Reproduction

by
Floyd E. Toole
Vice President Acoustical Engineering
Harman International

Part 2 – Making a good loudspeaker - Imaging, space and great sound in rooms.

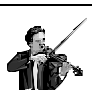
Loudspeakers can be designed to be "room friendly" so that they can sound good in a variety of different rooms. Controlling reflections can optimize imaging and spatial effects.

Some of what we think we know about audio is BS!
Before Science

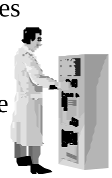
BS

2

Audio is an industry that is unfortunately burdened with a lot of ideas and beliefs that are not based on physical facts. Our business is complicated enough without adding the confusion of half-truths and folklore. An understanding of at least the fundamentals of room acoustics, and how loudspeakers and rooms interact with each other, takes us a long way towards our goal of truly excellent sound. This benefits everyone, the customer, the installer/consultant/retailer, and the manufacturers of the equipment.



Music and Movies
are Art



Audio is a Science

Science in the Service of Art
is our Business!!

3

The fact is that audio is a technology with a firm scientific grounding under most of it. It is not an art, and audio products do not possess artistic characteristics, except in the visual sense. The better we understand and use the real science to achieve excellent performance from our products, the more often the true audio artistry, the sound of music and movies, will be heard by us and by our customers.

The existence of custom audio installers and consultants is something relatively new to our industry. For the first time in the history of audio, there are professionals whose job it is to help the customer, in his or her own home, to achieve the best possible sound.

The Goal: To deliver high quality sound to our customers' ears.

The Problem:
ROOMS, the final audio component.

They affect sound quality and imaging
They dominate bass quality
They do this during the making of the recordings, and during their playback at home.
They are all different.

4

The traditional problem in audio has been that the room, the final audio component, is not within our control. Customer satisfaction, assuming that it is based on good sound, has been, therefore, a matter of chance.

This can change. With the selection of the appropriate loudspeakers, the application of some fundamental room acoustical knowledge and, if necessary, the right kind of equalization, we can greatly increase the odds in the favor of the customer and thereby ourselves.

Loudspeakers should sound good . . .

. . . and that is part of the problem.

How do we judge what is "good"?

5

No matter what measurements tell us, a loudspeaker isn't good until it sounds good. Complications in determining what is "good" include variations in rooms and recordings. The latter is something often ignored as we go about our daily businesses.

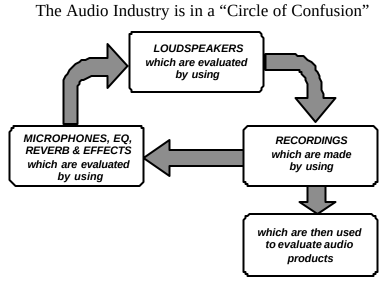
We listen, of course .

But when we do . . .

6

When we listen we are instantly trapped in the "audio circle of confusion".
Loudspeakers are evaluated by listening to recordings.
Recordings are made using microphones that are selected and positioned, equalized and processed in a variety of ways using the masses of equipment in a recording studio.
All of this is done while listening through loudspeakers in a room – a recording control room or movie dubbing stage.
The quality of the sound in a recording is very much dependent on the quality of sound from the monitor loudspeakers in that particular room.

The Audio Industry is in a "Circle of Confusion"



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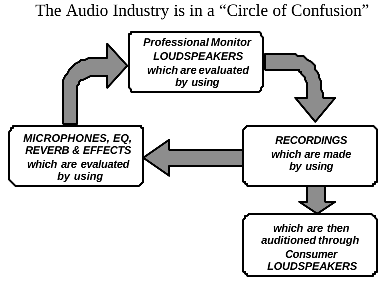
The recording industry has NO critical standards relating to loudspeakers for monitoring and for the rooms in which they are used. Consequently, recordings are extremely variable in quality, even in the gross characteristics of bass and treble balance. Yet, we try to evaluate audio products using such recordings. It is like making a technical measurement with an undefined test signal. The result is that mistakes are made. We cannot tell whether a good sound is the result of a truly good loudspeaker/room combination, or whether it is a case of compensating errors: a recording with, for example, too much bass being combined with a playback system that is deficient in bass.

The Audio Industry is in a "Circle of Confusion"



8

Some control rooms sound superb, while others are back in the dark ages of loud "mid-fi". Some even go out of their way to use bad monitor loudspeakers that they think represent what people are listening to in their homes and cars. It is obvious to anyone who listens carefully that all loudspeakers are getting better, and that the good ones are sounding more and more alike, and much more like the "real thing". However, bad loudspeakers can be bad in an infinite number of ways. No two are alike, and they can be dramatically different. How, then, is it possible for one "bad" monitor loudspeaker to represent the huge variety of sounds from clock radios, boom boxes, mini-systems, headphones and entry level car audio? It isn't!!

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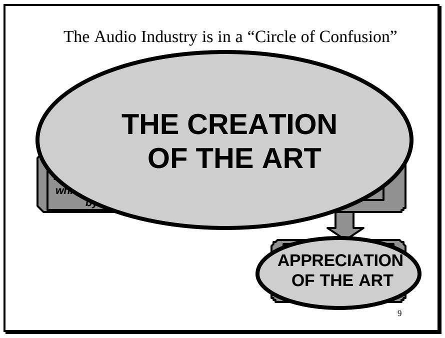
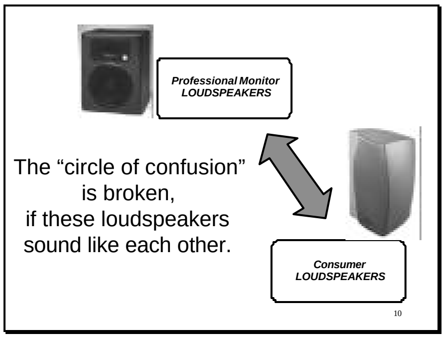
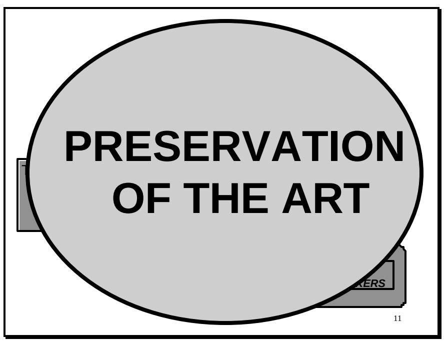


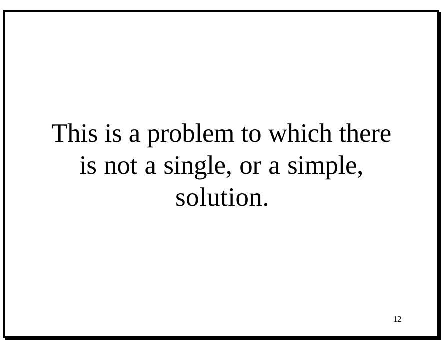
All of us need to exercise whatever influence we have to elevate the quality of sound everywhere. Then, and only then, will we have some assurance, when we listen at home or in our cars, that we are hearing what was intended by the artists. The enemy in this effort is ignorance and apathy. Most customers are intimidated by these kinds of decisions, and some truly say they don't care. Yet, I have never in my life demonstrated a truly good sound system to anyone who was not impressed, if not absolutely "blown away".



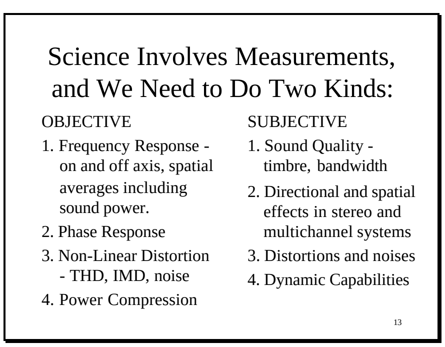
Ironically, the problem exists at both the professional and consumer levels. Both need to be aware of the genuine advances in acoustic science and technology.



Only then can we say that we are working within an industry that aims to preserve the audio artistry.



If we cannot totally rely on our ears, what else is there?



The scientific method requires data. Data of all kinds, and the more the better. In this case, we must use technical measurements, because they are the essential tools of the engineers designing the products. It is necessary to measure everything that we think might be relevant to how something sounds. This is more than is commonly thought. However, we also need subjective data, relating to listeners' opinions of the many perceptual dimensions of sound quality as well as spatial and directional attributes.

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Scientific Design Requires:

(a) carefully controlled listening tests: subjective measurements

combined with

(b) accurate and comprehensive technical measurements

combined with

(c) knowledge of the psychoacoustic relationships between perceptions and measurements .

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If listeners like or dislike something, it is important to try to identify what, technically, was responsible. By taking the listening reports back to the lab, it is possible to learn the relationships between what we measure, and what we hear. This is the science of psychoacoustics, and the better we understand it, the better we will be at delivering good sound to our customers' ears.
Technical measurements must be accurate, or else they lie to us. Getting accurate acoustical measurements without good facilities is very difficult to impossible. Much of the data floating around the loudspeaker industry is not accurate.

What are Subjective Measurements?

They are LISTENING TESTS in which the OPINIONS OF LISTENERS relate to the SOUNDS OF THE LOUDSPEAKERS rather than:

How they look
How much they cost
What reviewers think of them
What listeners think of the manufacturer
How loud they are
How the room is physically arranged
Etc. . . .

These are NUISANCE VARIABLES -they MUST be controlled

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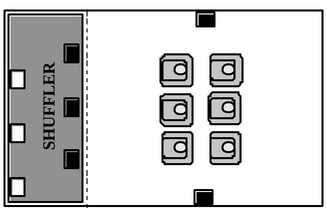
Most listening tests are valid only at a specific time and place, for the specific recordings that were listened to, and for the people offering their opinions. This can be acceptable if it is you choosing your own system in your own home. It is not acceptable for a loudspeaker manufacturer, who is trying to design products that can sound good to many listeners, in many rooms, with many recordings. Consequently, we get scientific about it, and start to remove some of the variables that have nothing to do with the sound from the loudspeaker, letting the listeners focus as much of their attention as possible on the sound, and the sound alone. Purveyors of 'magic' in the audio industry find these double-blind tests very threatening.

Mono, Stereo and Multichannel Tests



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One of the most problematic nuisance variables is the position of the loudspeaker and the position of the listener in the room. To control this variable at Harman, we have created a room with a "shuffler" that physically moves speakers around, bringing them always to the same locations when they are being auditioned. It is pneumatically driven, quiet, and computer controlled so that the positions are precise. It takes about three seconds to switch the positions of the active and inactive loudspeakers.

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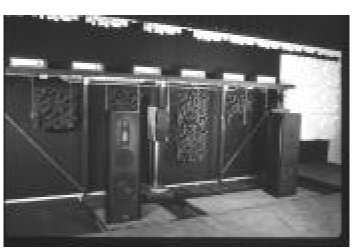
A pair of speakers ready for listening.



18

And, about three seconds later, a second pair can be heard.

What the Listener Sees

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The listener (and we prefer to use one listener at a time) sees none of this, of course. Here we show a video display on a large perforated screen. For the evaluation of most products this is not used.
The tests are controlled by the listener, who takes as long as is needed in order to form a satisfactory judgment. A computer randomizes the choice of music, and the coded identity of the test loudspeakers for each musical selection, so that the opinions must relate as much as possible to the sound itself.
Listeners are selected for normal hearing and aptitude, and then are trained to be really fussy. They yield remarkably consistent opinions.

In the Real World

The room is the final audio component,
and it is not under control!

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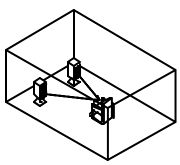
"Where the rubber hits the road", in the customer's home, we have no such conveniences, so we must develop products and techniques that allow good sound to prevail even when the local acoustical conditions are less than ideal.

This is where knowledgeable custom installers, consultants and audio specialists come to the rescue.

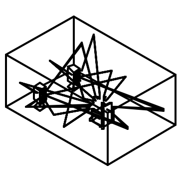
The Room - Direct Sound



21

The first sound to arrive at a listener's ears is the "direct" sound. If the loudspeakers have been angled to face the listener, this will be the on-axis sound, often the best possible sound from the loudspeaker.

The Room - Early Reflections



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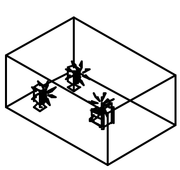
However, following only a few milliseconds behind, and only slightly less loud, will be the early reflections: sounds that have been reflected from only one surface in the room.

The Room - Reverberation



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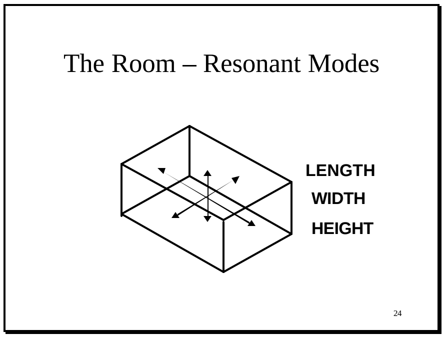
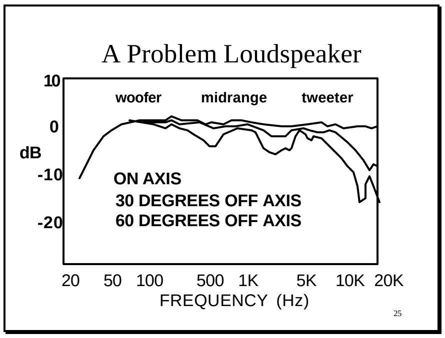
Still later, come the multitudes of reflections that have been reflected more than once, perhaps many times. These are individually much lower in amplitude, but collectively loud enough to be a powerful factor in our impressions of sound quality, space and imaging. In small rooms, typically furnished, this sound field, although often called 'reverberation' is not the directionally diffuse and temporally complex reverberation that we hear in a concert hall, or many other large, acoustically 'live' spaces. Some would argue that it deserves a different name.

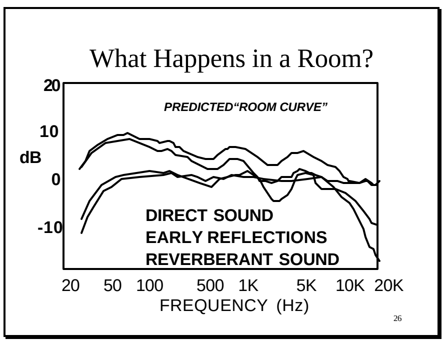


Rooms also have resonances that emphasize certain frequencies, attenuate others, depending on the dimensions and shape of the room. And, they do so in a manner that depends on where the loudspeakers and listeners are located within the room. These effects are strongest at low frequencies.

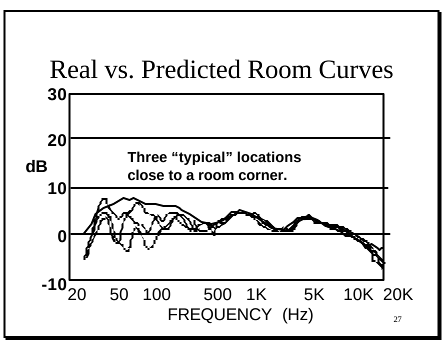


Using a loudspeaker that we know, in retrospect, had a design flaw, let us see what happens in a room. This loudspeaker was designed using the philosophy that the direct sound, the on-axis sound, is the most important. The top curve is the on-axis measurement, and it is very smooth and flat, a credit to the design team.

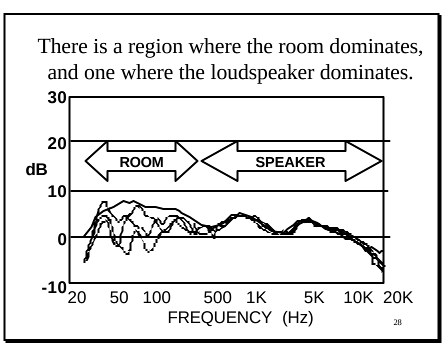
The second and third curves, moving downward, are the 30- and 60-degree off-axis measurements, showing that these sounds are not nearly so neutral; the output varies with frequency. What happens to this in a room?



The data in this slide are derived from many measurements made in a large anechoic chamber. This is a room having no "echoes", used for acoustical measurements. All surfaces are covered with highly effective acoustical absorbing material that is, in this case, about four-feet thick. The color coding of the curves is not visible here, so it is not possible to see which curve is which. However, one might recognize the flat on-axis curve representing the direct sound, and see also that none of the other curves is remotely smooth or flat. This tells us that all of the sounds arriving at the ears do not convey the same message about the sound quality, or timbre. The top curve is a calculated prediction of what a measured "room curve" might be.

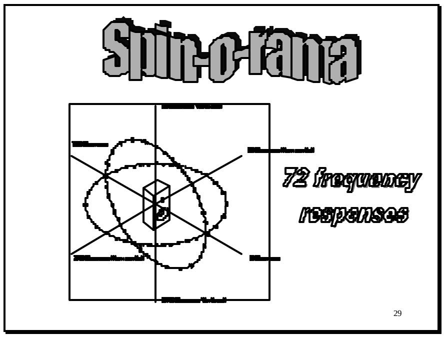


The loudspeaker was then placed in a typical left or right channel location in a real room. It was measured at the listening position, then moved to two other locations within a radius of two feet, and measured at each location. The fourth curve, the top one, is the calculated room curve from the previous slide. Obviously, little is changing at frequencies above about 300-400 Hz, and the prediction is right on target. However, below these frequencies, there are considerable location-dependent changes, and the prediction fails completely. The reason? Room resonances and boundary effects that are specific to that particular room. These can only, with precision, be evaluated by measurements in the room itself.

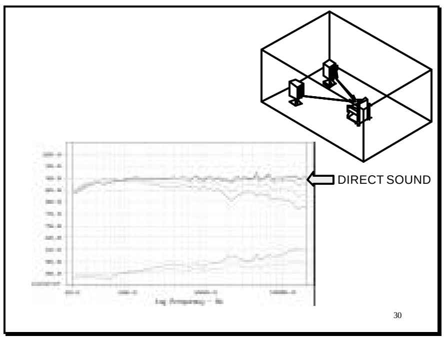


However, it is also clear that through the middle and high frequencies the anechoic measurements made in the laboratory have done an excellent job of predicting what happened in the room. However, doing so required many, many measurements at positions all around the loudspeaker.

If we are to try to anticipate how a loudspeaker will sound in a room, it is necessary to measure everything, and not just a few curves around the on-axis measurement.

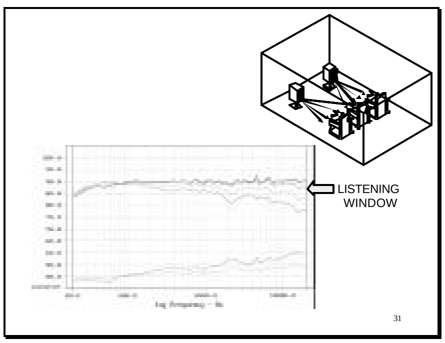


At Harman, the engineers have dubbed our basic measurement of frequency response, the Spin-o-rama, since it involves spinning the loudspeaker on two axes, and accumulating a total of 72 measurements.

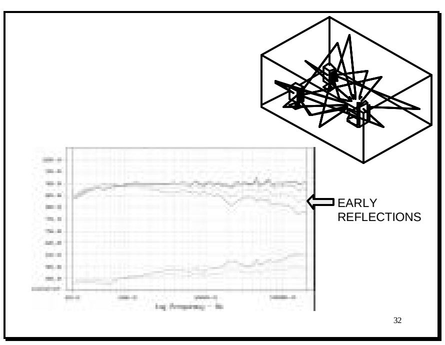


The collection of raw data is computer processed to generate a set of curves showing estimates of the distinctive regimes of sound arriving at a listener's ears in a typical room. To do this, a large survey of real rooms was undertaken, and a statistical analysis of angles and distances led to the algorithm that generated these curves. All measurements have a frequency resolution of 1/20 octave, from 20 Hz to 20 kHz.

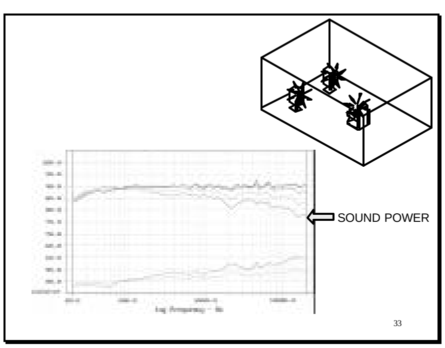
The top curve is the on-axis curve, representing the direct sound for a person in the "sweet spot".



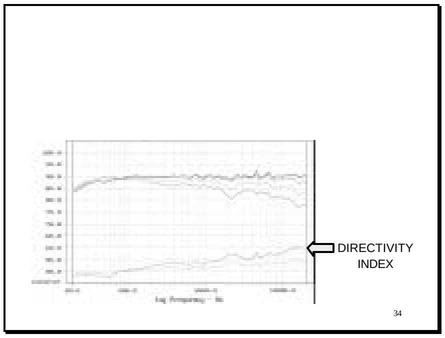
The second curve is a spatial average over +/- 30° horizontal, and +/- 10° vertical, representing the direct sound for listeners seated in a row of chairs or a large sofa, and possibly standing and sitting.



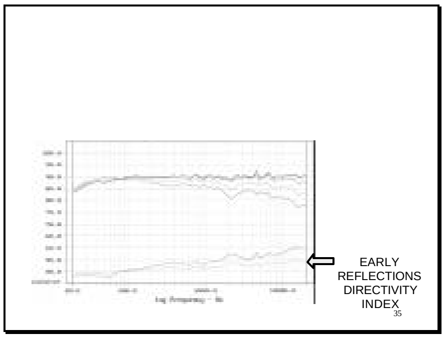
The third curve is the energy sum of the set of early reflections. Ideally, these should look a lot like the on-axis curve, so that it conveys the same timbral information.



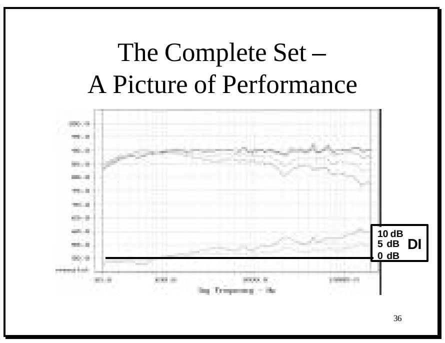
The fourth curve is a calculation of the total sound power radiated by the loudspeaker in all directions (this is NOT a simple average or sum of all 72 measurements). Again, this curve should be smooth and flattish.



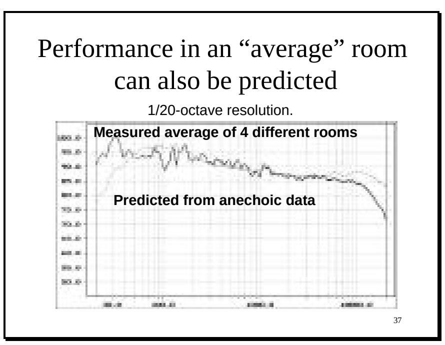
The uppermost of the bottom pair of curves is the Directivity Index, or DI. This is an indication of the angular uniformity with which the loudspeaker radiates its energy into a room as a function of frequency. It is a measure of the uniformity of its dispersion as a function of frequency.



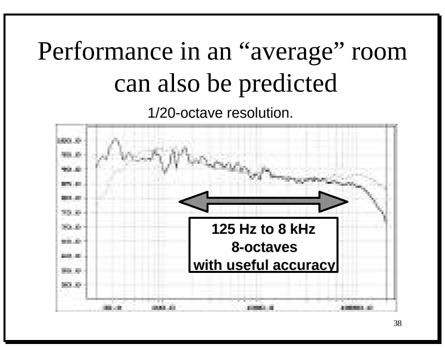
The bottom curve is an invented DI, this time just for the early reflections.

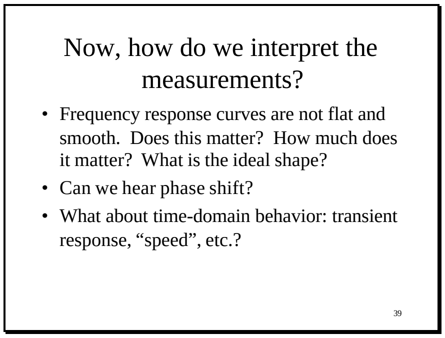


Here we see the complete set as they are presented for visual inspection. The whole idea of this is to present to the eyes, a set of data that can be interpreted in a way that allows one to anticipate how a loudspeaker might sound in a room.
The curves shown here describe a truly excellent loudspeaker, not perfect but, currently, a good example of the state of our art. Note the smoothness of all of the curves, and the basic similarities in all of the curves, from the single on-axis measurement, through to the estimate of the total sound radiated in all directions, the sound power.

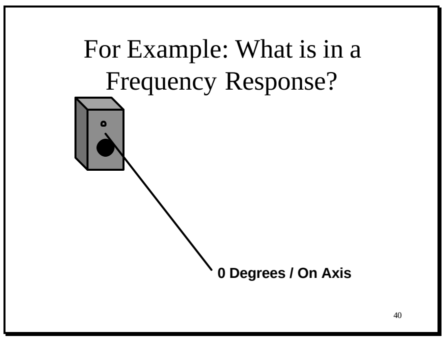


In the same way that the earlier example was calculated, we can generate a curve that tries to anticipate a room curve. Here we have not included all of the possible refinements, but it suffices to get us within a couple of dB over most of the frequency range.

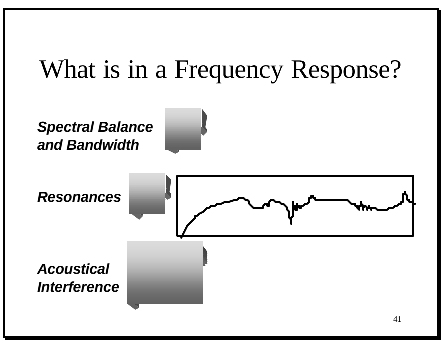




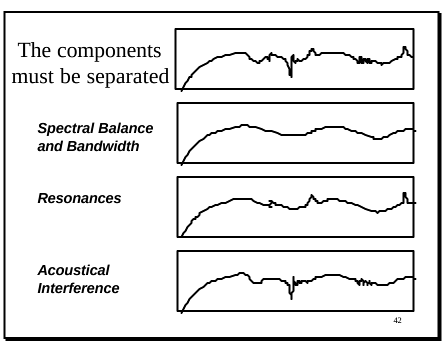
O.K. So we get some curves. The real problem is that they ARE curves, and not straight lines. What is the ideal shape? How much deviation from the ideal is audible? Is there more to this than just frequency response?



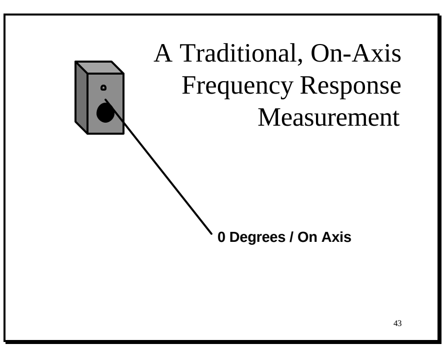
Let's start with the most basic of all measurements, the frequency response. In the case of a loudspeaker we would begin with a look at what happens on the major axis.
Incidentally, such a measurement should be made at a distance of 2 m (6 feet) or more. The industry standard specifies loudspeaker sensitivity at one meter, however, the standard also requires the measurement to be made in the "far field" of the source, and if necessary, for the measurement to be calculated back to 1 meter. Many people mistakenly do not do this, and also make frequency response measurements at 1m. For loudspeaker systems of typical size, these measurements can exhibit large errors.



Of the features that our eyes can extract from a curve like this, it is obvious that spectral balance and bandwidth are important. Try playing with the bass and treble controls, and you will find that small changes are audible.
Resonances are REALLY important because our perceptual system (the ears and brain) is highly sensitized to them. The reason: resonances are the "building blocks" of all of the sounds that we are really interested in listening to – voices and musical instruments. Resonances can cause peaks and dips in a frequency response curve. However, so can acoustical interference, a phenomenon that turns out to be much less audible under normal listening circumstances.



So, we need a measurement system that allows us to separate, visually, those features in a curve that are caused by each of these phenomena. Only then can we be truly analytical, and make good judgments about how good or bad the device is.



Once upon a time, it was thought that a single curve told us useful information.

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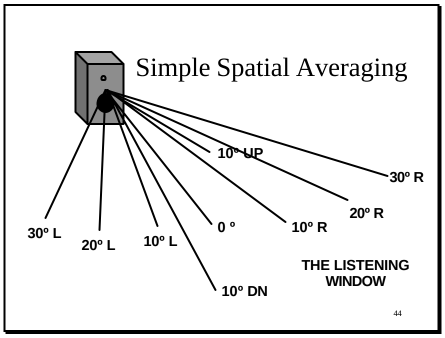
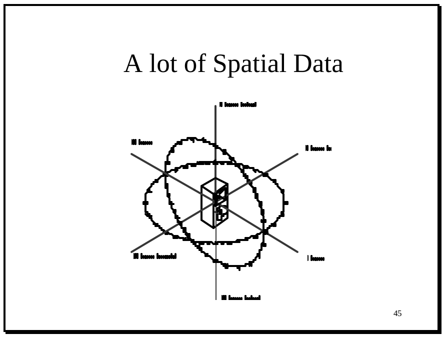


Then we learned that spatial averaging allows us to separate those peaks and dips caused by resonances from those caused by acoustical interference. The explanation is really simple: those features associated with resonances tend not to change when the microphone location is changed, while those associated with interference do.



When we average a lot of measurements made at a lot of different locations, and certain visual shapes do not disappear, we can be quite confident in concluding that those are resonances, and not the result of acoustical interference.

Spatial Averaging in Rooms

- Helps to reveal the presence of resonances, which <u>can</u> be equalized.
- Attenuates the effects of acoustical interference, which <u>cannot</u> be equalized.
- Helps to eliminate visual evidence of dips in the frequency response, and thereby the temptation to try to fill them with equalization – which does not work!

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In rooms, there is an abundance of acoustical interference, caused by multitudes of reflections. Therefore, spatial averaging, i.e. combinations of measurements made at several locations, can help to isolate resonances. This is important because it turns out that we can equalize resonances (about which, more later), and we cannot equalize the effects of acoustical interference.

Resonances are major problems!

- In loudspeaker drivers
  - cone flexure modes
  - suspension and frame modes
- In enclosures
  - mechanical resonances in panels and surfaces
  - acoustical resonances in cavities
- In rooms

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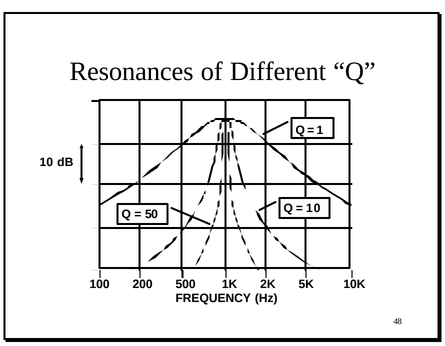
So, we expend a great deal of effort eliminating resonances from loudspeaker systems, and when they are installed in rooms, we need to spend some time and effort to identify and eliminate serious resonance problems.



Resonances are differentiated by their "Q", or quality factor. A high-Q resonance is one that is very frequency specific and that rings a long time. An example of a high-Q resonance is an empty wine glass, held by the stem, and tapped with a finger nail. It emits a clear tone that rings. If one places a finger on the side of the glass and taps it again, the ringing is shorter. The finger has taken some energy out of the resonant system, and the 'quality' is reduced. If the entire glass is grasped by a hand, and the tap is repeated, there is almost no ringing at all. A tone is still recognizable, but it is a low 'quality', or low-Q, resonance. High-Q resonances have sharp peaks, and low-Q resonances are broader when they are seen in frequency responses.

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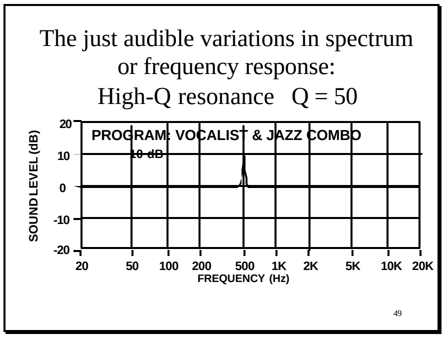
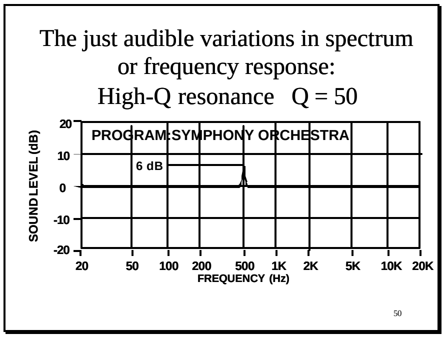
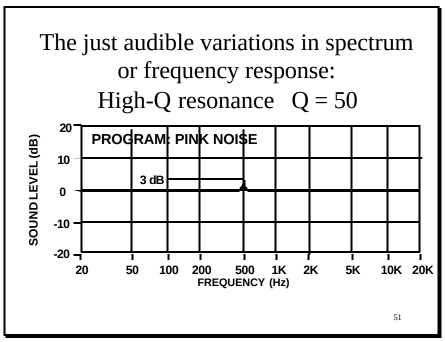


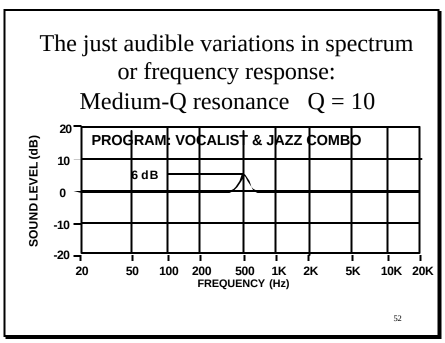
Sean Olive and I, when we were at the National Research Council, in Canada, published a paper in which we showed the shapes of deviations in frequency responses that corresponded to the just audible thresholds for resonances of different Q, at different frequencies, for different kinds of music or sounds. The effect of frequency was secondary, so here I show only what happens at 500 Hz. The results at different frequencies are similar. It shows that, for multimiked pan-potted, low reverb, pop or jazz, the threshold of audibility corresponds to a 10 dB spike in a frequency response curve. It looks bad, but it is just barely audible!



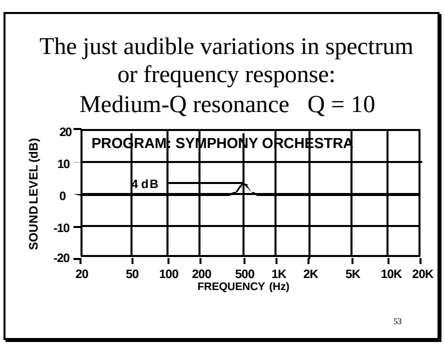
With a big band or symphony orchestra (complex orchestration) in a reverberant hall, the threshold is lower (we are more sensitive).



Of all the signals we tested, pink noise was the most revealing of resonances. It produced the lowest thresholds. Such low-amplitude, narrow, spikes are difficult to measure with precision at all frequencies.



When the Q is reduced, the pattern of audibility is much the same, but the thresholds are lower.



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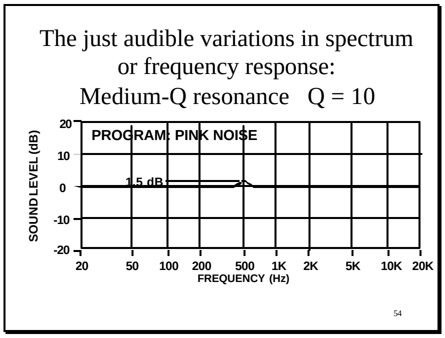
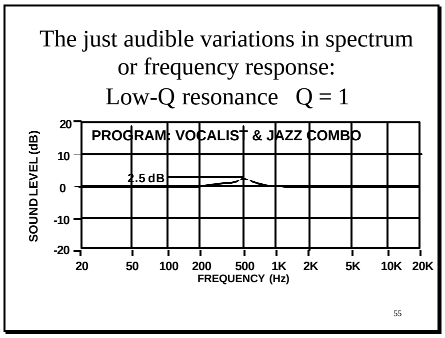
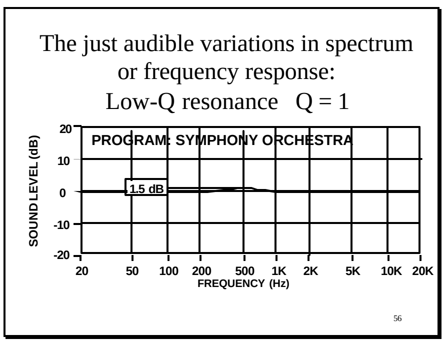
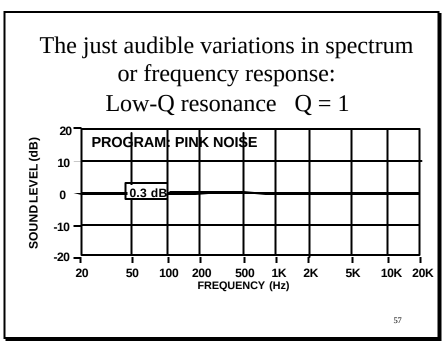


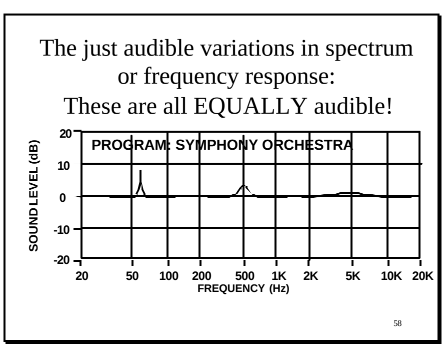


When we get to really low-Q resonances, the ones that ring very little, it turns out that we can hear them at very low measured amplitudes. What, then, of the arguments that the ringing of high-Q resonances "smears" sounds, making them less articulate? These are arguments that are most likely based on visual interpretations of measured data, not on actual subjective tests of the audibility of the effects. They sound as though they should be true but, except at very low frequencies, they are fanciful. Good engineering should attempt to eliminate resonances of all kinds, but it is important to understand what is and is not audible.





This "curve" looks almost like a straight line. Our eyes are telling us that it is almost perfect, yet our ears are telling us that there might still be something audibly wrong.
So, in this case, what our eyes tell us does not intuitively correspond with what we hear. This is why it is so important to do the science, and to establish what the real psychoacoustic relations are. Our instincts can be wrong.



Is there an explanation? It is probably because music and speech are ever-changing. Also, voices and many musical instruments are played with vibrato – a modulated pitch. High-Q resonances take time to build up, as well as to decay. We tend to talk about the ringing, overhang or decay of resonances after the signal has stopped, ignoring the front-end effect. High-Q resonances are narrow, very frequency-specific, and musical sounds must be sustained long enough to energize them. Few are. Low-Q resonances are wide enough that they respond to everything, and they take almost no time to reach full amplitude.

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Measurements must have enough resolution to show what we can hear

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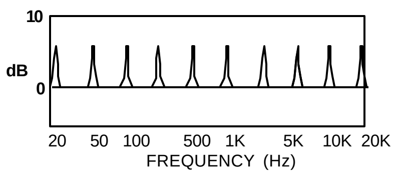
In order to make any sense at all of a frequency response curve, or a set of curves, they must be capable of revealing to our eyes everything that is audible.

The belief, still widespread in this industry, that we cannot measure what we can hear, has its origins in situations where the measured data were erroneous or incomplete. Such situations are common in the loudspeaker business.

A Resolution Test

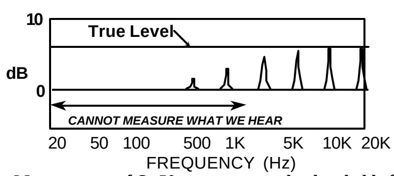


Q=50 resonances at the threshold of audibility

60

Let's create a test. Suppose we had an imaginary system in which there were high-Q resonances uniformly distributed from low to high frequencies. A competent measurement system would reveal them to our eyes as they truly are.

A Partially "Blind" Measurement

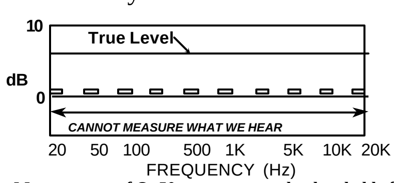


Measurement of Q=50 resonances at the threshold of audibility using a time window of 17 ms (60Hz)

61

However, not all measurement systems are equal. Many very commonly used ones "gild the lily", making the curves smoother than they really are. All systems that use time windowing, or the equivalent (MLSSA, TEF, and any FFT-based system) can do this IF the measurement window is not sufficiently long. Here I show what happens with a quite long window (17ms), more than is used by many manufacturers and reviewers. It is clear that the measurement does not reveal the existence of the high-Q resonances in the middle and low frequency regions. It does not show things that we know we can hear.

A Totally "Blind" Measurement



Measurement of Q=50 resonances at the threshold of audibility using a 1/3-octave analyzer

62

The popular one-third-octave measurements, very common in room measurements, simply fail. These give only a very "broad-brush" view of what is happening, and are of very limited use. One needs an analyzer capable of at least 1/10 octave resolution in order to reveal what we need to see.

A 4π (full space) anechoic chamber

4-foot long wedges: anechoic to 60Hz, calibrated to 20 Hz

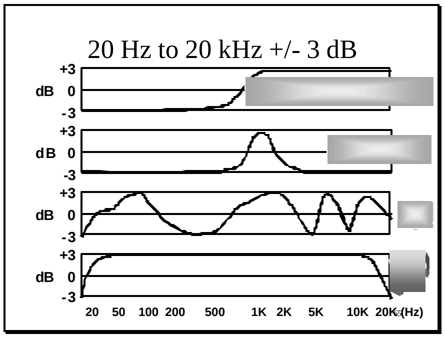
Automatic positioning: motorized, computer-controlled platform

For loudspeaker measurements, very long measurement windows are necessary, and these can be accomplished only in anechoic spaces. Outdoors, away from all reflecting surfaces, is free, but impractical. Anechoic chambers, such as the one shown here, are very practical, but also very expensive. However, this is the price of entry if you are seriously in the business. The length of the wedges determines how low in frequency one can measure accurately. These 4-foot wedges create a reflection-free environment down to about 60 Hz. We have calibrated it down to 20 Hz for specific measurement locations within it. With a large enough measurement time window, ANY measurement system should yield accurate data.

AND the most commonly used specification for frequency response is useless . . .

. . unless it is accompanied by a graph!!!

64



Every audio device has a specification for frequency response. A tolerance of +/- 3 dB is sufficient to describe a range from junk to jewels. By itself, it is meaningless window-dressing. A curve, and the ability to interpret it, are necessary. If the tolerance is small enough, then it does have meaning, of course.

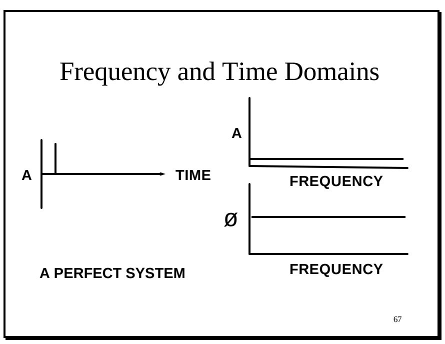
Frequency and Time Domains

- Related by Fourier Transformation
- Behavior in one domain is paralleled by corresponding behavior in the other

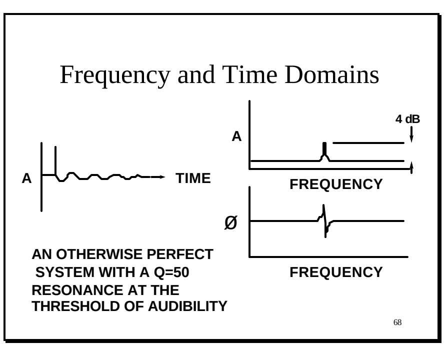
66

So far, we have talked about frequency response as though it were the only important factor. What about the all-important transient response, speed, punch, drive, and all of those descriptors of what happens in the time domain?

Well, it turns out that the two domains are related to one another, by the Fourier transformation.

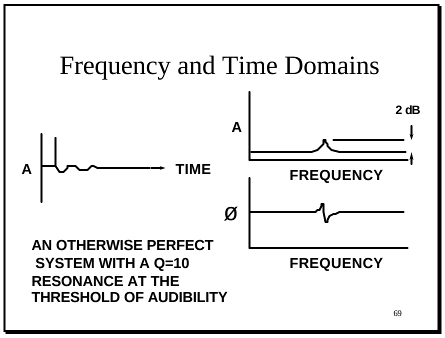


A perfect linear system would be described either by a clean uncluttered transient, or by a pair of flat straight lines portraying a constant amplitude vs. frequency characteristic (we call this the frequency response, although it is really the amplitude response), and a constant phase vs. frequency response. The flat amplitude response tells us that the signal level at all frequencies is constant. The flat phase response tells us that everything is happening at precisely the right instant in time. The combination of flat amplitude and phase responses correspond to a perfect impulse, or transient, response.

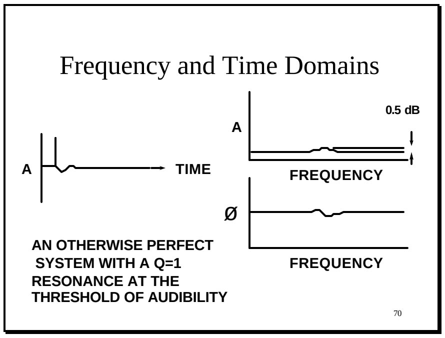


Here we have disrupted the perfect system with a single high-Q resonance. The narrow “footprint” in the amplitude response, as seen earlier, is repeated in the phase response. In the time domain, the corresponding effect is extended ringing (the empty wine glass).

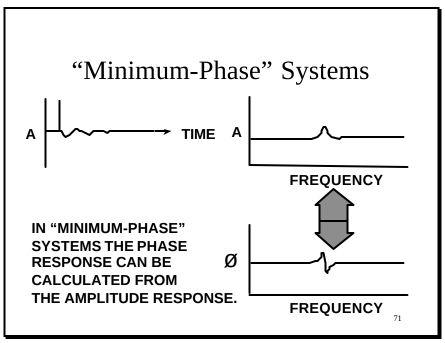
If we measured the amplitude and phase responses, a computer could perform a Fourier transform and give us the transient response. If we measured the transient response the computer could calculate the amplitude and phase responses. So the information on the left side of the slide is the same as that on the right side, only displayed in a different form.



Here is a medium-Q resonance. The frequency-domain 'footprint' is larger, and the time-domain 'footprint' is smaller.



And a low-Q resonance. Note the convenient relationship: as the footprint in the frequency domain gets larger, that in the time domain gets smaller.



There is a class of systems that behave as "minimum-phase" systems. In such systems, if one has measured only the amplitude response – our familiar "frequency response", it is possible to calculate the phase response from that data. Now, if we know both the amplitude and phase responses, we can calculate the time response. So, in a minimum phase system, a measurement of the frequency response, allows us to predict the time response. A bump in the frequency response means that the system must ring. A flat, smooth, frequency response means that there is no ringing. The previous data show that we are able to measure the visual evidence of audible resonances in frequency response curves. This is really important.

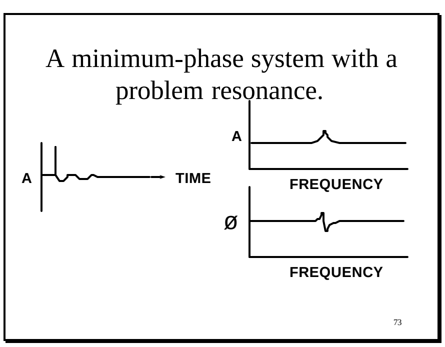
What Systems Exhibit Minimum Phase Behavior?

- Many common functions in analog and digital electronics.
- Loudspeaker transducers – conventional woofers, midranges and tweeters.
- Room resonances at low frequencies.

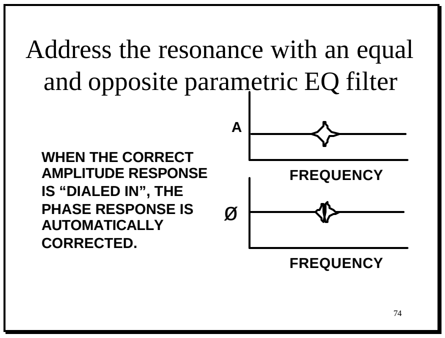
This is a HUGE advantage!

72

Several very important devices are minimum phase systems, meaning that, for these devices, the frequency response curve is the single most important measure of audible performance in the linear domain. Of course we do measurements of non-linear effects as well, but in general these are much less troublesome.

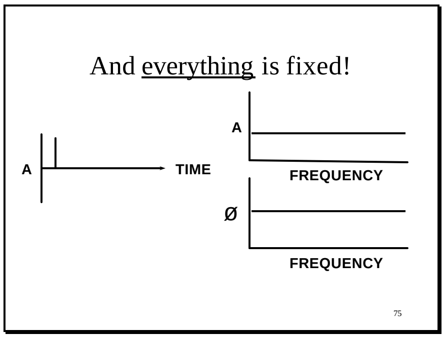


If a minimum phase system has a resonance, and we wish to get rid of the audible effects, we can choose to do it electronically.



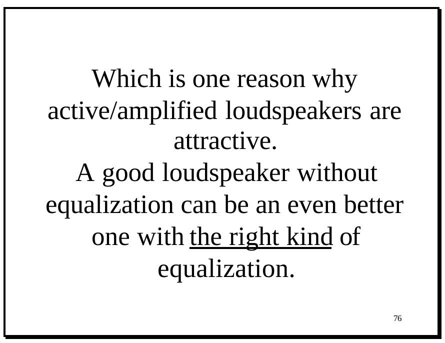
Simply design a minimum-phase filter, in either analog or digital electronics, that exactly matches the shape of the bump in the frequency response, but is inverted. When the two are added, we get a straight line. The filter, because it is minimum phase, will have a phase shift that mirrors the phase shift in the resonance, so that a summation yields another straight line.

How often have you heard that equalizers are bad because they add phase shift? Here we show that it is a good thing – assuming that it has all been done properly, with the necessary precision.

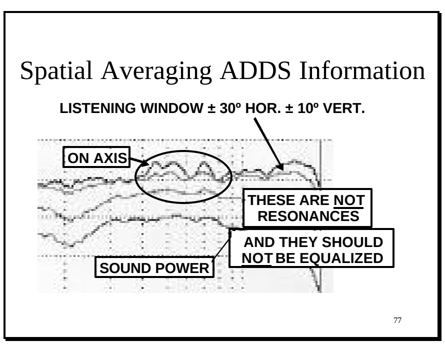


Two flat lines on the right, we know, correspond to a perfect impulse response on the left.

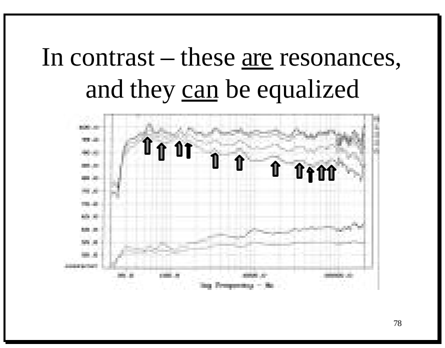
This is a very simple form of pre-distortion, a well known technique that, with the advent of digital processing, is likely to become more widespread. If we know what an electromechanical or acoustical system is doing wrong, there are some things that can be corrected by modifying, or pre-distorting, the signal so that what is eventually radiated as sound is correct.



It is not magic, but it certainly seems like it. Good loudspeakers can be made better. Room resonances can be tamed (for specific listeners at least).
However, in order for it to work we need accurate, high resolution, frequency response data, and parametric filters.



Which brings us back to measurements and spatial averaging. Here we see a portion of a spin-o-rama for a loudspeaker showing some bumps in the on-axis curve. The bumps are attenuated, and even disappear, with spatial averaging. This tells us that the bumps are caused by acoustical interference (in this case diffraction from the cabinet edges). They are not resonances, and they should not be equalized.



In this example, the series of bumps penetrate all of the curves, surviving even the 72-curve calculation of sound power. These truly are resonances, and they can be treated with individually designed parametric filters.

Spatial averaging ADDS information. Spectral averaging (smoothing) takes it away.

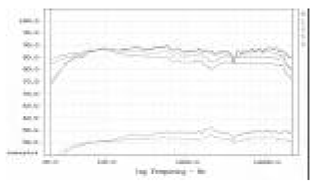
Measurements make a nice story, but can people really hear the differences?

Let's test them with four "high-end" speakers.

79

If all of this really means anything, we should be able to prove it using our carefully conducted listening tests.

This one looks good: $10,000/pr

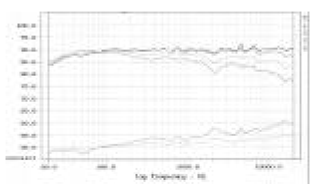


80

Four expensive and highly regarded loudspeakers are evaluated in the "shuffler" room, in a double-blind test. These are the measurements. The listeners, of course, do not get to see them until it is all over.

This one is well behaved. It has smooth, flattish curves, very wide, very uniform dispersion, and excellent low-frequency extension. The slight sag in the upper middle frequencies is something that is sometimes done to compensate for the numerous excessively bright recordings out there. The trade-off, others might sound a bit laid back.

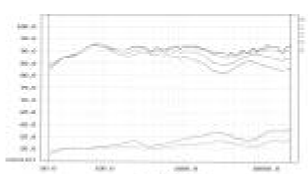
This one too: $8000/pr



81

Here is a 'tell it the way it is' speaker. Unabashedly flat, very smooth, superb low bass extension, being only about 5 dB down at 20 Hz, with a directivity that smoothly and gradually rises with frequency. A small dip in sound power around 2 kHz might be barely audible, but likely only in quite 'live' rooms.

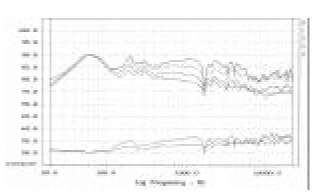
A bit wobbly: $8000/pr



82

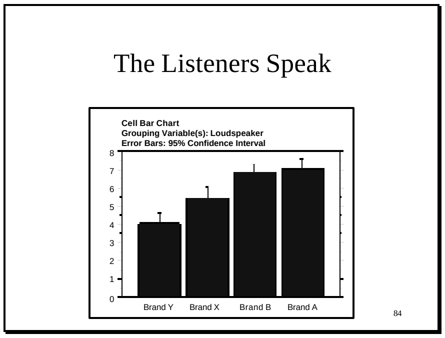
This one is likely to have a 'personality'. The undulations in the upper mids/lower treble are everywhere, including the directivity. It is possible to play detective, and guess that this is a three-way system, with the woofer crossing over to the midrange around 300 Hz, and the midrange crossing over to the tweeter around 3-4 kHz. How do we know? Look at the directivity curves. Very low frequencies are omnidirectional. The curve rises as the woofer becomes more directional until it crosses over to the smaller midrange when the directivity drops. It then rises again with frequency until it crosses over to the small tweeter, at which point the cycle begins again. The low bass is fair, but a small bump just below 100 Hz spoils it.

A Mountain Skyline: $5000/pr

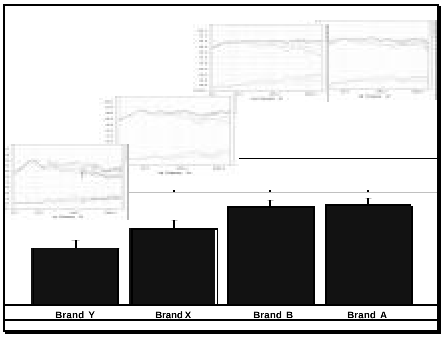


83

This speaker has a bunch of things going on. Clearly the designers didn't believe that flat was necessary, or they didn't know how to achieve it. Not only are the general trends not flat, but superimposed are peaks and dips suggesting resonances. The proof that they are resonances is in the fact that the patterns are repeated in all of the curves. The directivity is interesting, being zero up to 100 Hz (the woofer) and then abruptly rising to about 5 dB and hovering around that all the way to 20 kHz. Since 4.8 dB is the directivity of a dipole, we could suspect that this is a hybrid system with a panel loudspeaker operating above about 100 Hz. The woofer exhibits a significant bump and then rolls off below about 60 Hz. No subwoofing here.

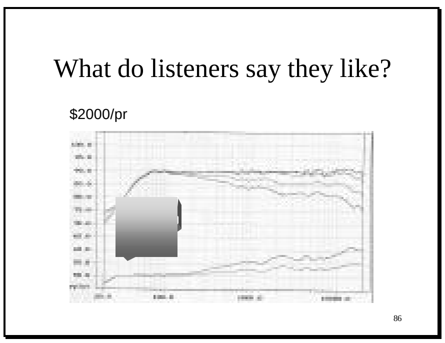


After several rounds of listening to different kinds of music, several listeners yielded subjective preference ratings that were processed in a statistical analysis program. One of the results is a bar graph showing the average rating for the group of listeners, for each of the loudspeakers. The tiny lines on top of the bars show the 95% confidence intervals. If the differences in the ratings are greater than these lines, the differences are probably statistically significant, and not due to chance. The two top-rated speakers are not significantly different from each other, according to this rule. The other two are truly less good.



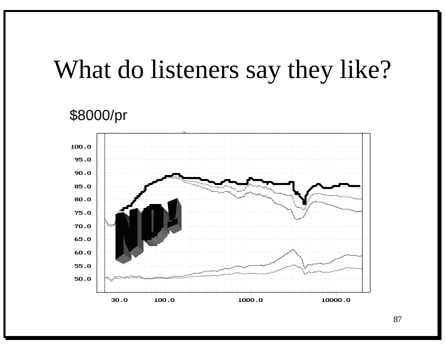
When we combine the subjective with the objective data, it is clear that the loudspeakers that yielded the best set of technical data, also were preferred by the listeners.

It works!

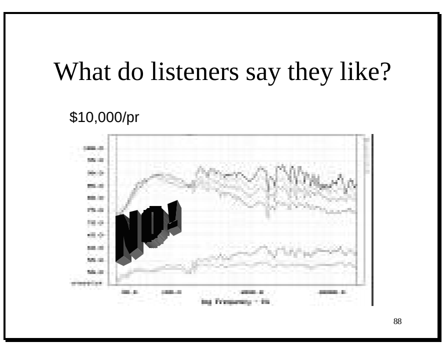


At Harman, we do hundreds of such listening evaluations, using competitors' products that we purchase on the open market. It is essential to know where our new products stand with respect to the competition.

The results are monotonously the same. Loudspeakers that look good in the spin-o-rama measurements are the ones that are subjectively preferred.



This is one of those pricey 'high end' bookshelf-sized speakers that some reviewers have raved about. The measurements suggest that it is a slightly dull sounding, moderately colored system with no real bass. The listeners agreed.



Note that price sometimes has nothing to do with sound quality. There may be material value to justify the high price, a sexy appearance, or just a lot of hype. This is the audio business, and reason and the laws of physics seem not to be universally applicable.

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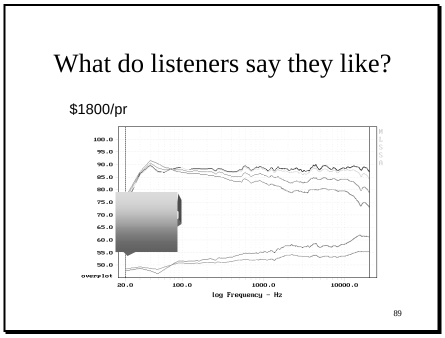
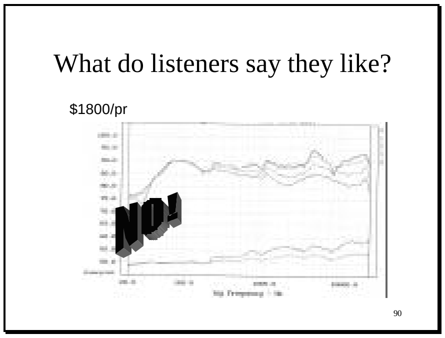
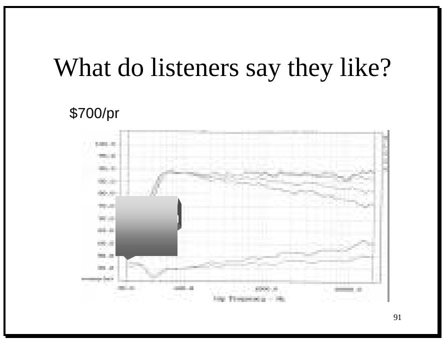
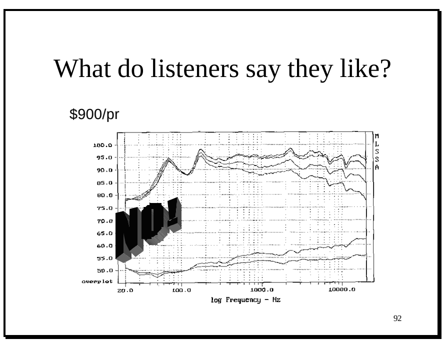
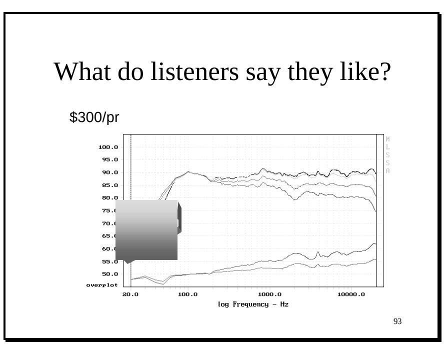


Truly excellent sound can be found at moderate prices.





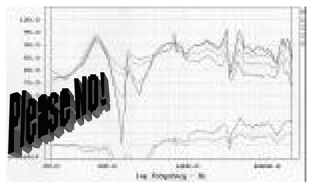




Even entry-level products can sound good. They will come in plain enclosures, and they may not play as loud as the bigger boys, but they are eminently capable of preserving most of the artistic integrity.

What do listeners say they like?

$460/pr



94

Here's one to avoid! It is amazing that anyone, especially a well-advertised brand as this one is, would actually let something like this out into the marketplace. For the same price, they could be selling good sound. Obviously they don't care. Instead they sell a slick package and a story.

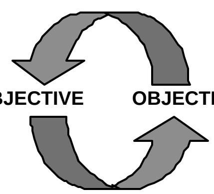
Conclusion:

Listeners don't like resonances!!

95

ALL of the most preferred loudspeakers are ones that exhibit the flattest, smoothest families of curves.
They exhibit the fewest, and the lowest level, resonances. They have the flattest, smoothest, widest bandwidth frequency responses when measured from all angles.
They have similar shapes in all of the curves – i.e. they have quite constant, or at least smoothly changing, directivity as a function of frequency.
Can we measure what we can hear? No, but we sure have made a good start.

Equivalent Measurements



96

This is a powerful position to be in, when it is possible to demonstrate that the right set of accurate measurements has a consistent relationship with listener evaluations.

We do not claim to have mastered everything, at this stage. However, some things are understood. They even make logical sense.

So, let's assume that, to a first approximation, we understand how to design loudspeakers that have the potential of sounding good in a room.

The Rules for Good Sound in Rooms

At middle and high frequencies:
- ✓ Start with a loudspeaker that was designed to function well in a variety of different rooms.
- ➢ Use geometry, reflection, diffusion, and absorption to achieve good imaging and ambiance.

At low frequencies:
- ➢ Maximize the output from the subwoofer(s).
- ➢ Achieve a uniform performance over the listening area.
- ➢ Equalize to achieve good performance.

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The second rule for good sound requires that we look at some specifics of the room itself.

Reflections alter both Sound Quality and Imaging

Reflected sounds can be controlled by:
(a) controlled-directivity loudspeakers,
(b) absorbing or diffusing objects on reflecting surfaces in the room,
(c) the shape of the room,
(d) some of each.

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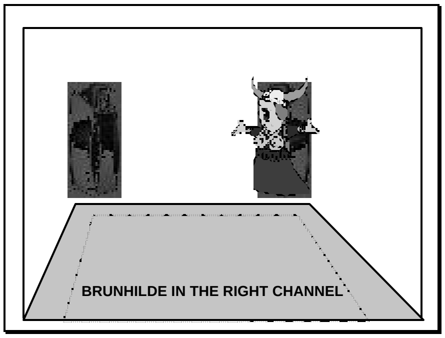
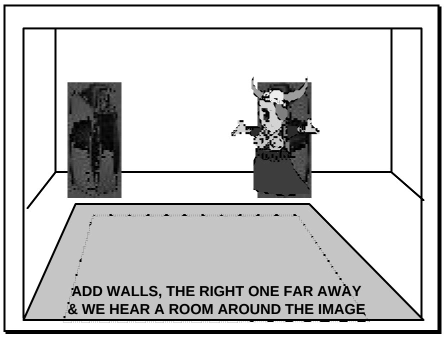
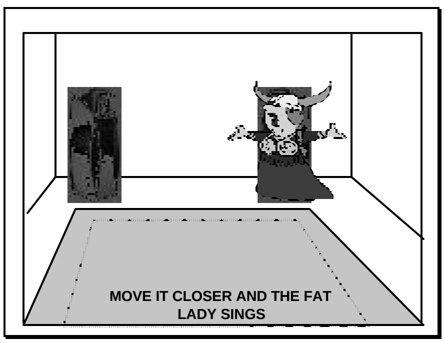


Here is a cartoon description of what happens with reflected sounds in a room.

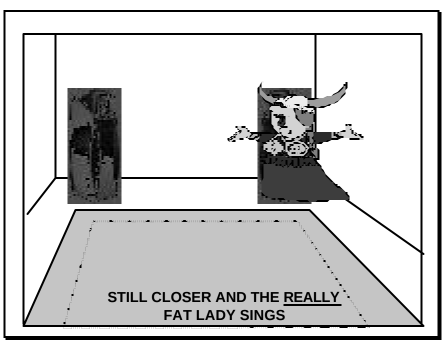
We start with only a floor, no walls. Brunhilde, of opera fame, is singing in the right speaker only.



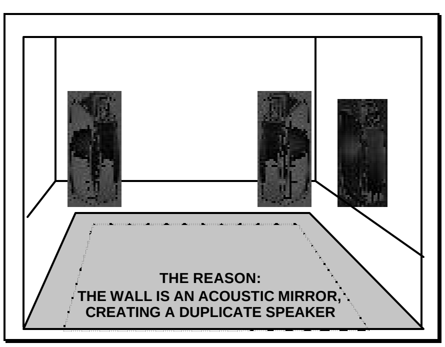
Adding walls, the one next to the right loudspeaker is some distance away, produces a nice warm "spatial" illusion. It sounds a little richer.



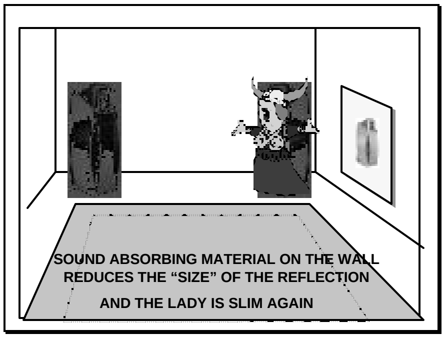
If the wall is moved a bit closer to the speaker, we note that the lady is a little bit "smeared", putting on some weight, and maybe leaning a bit to the right.



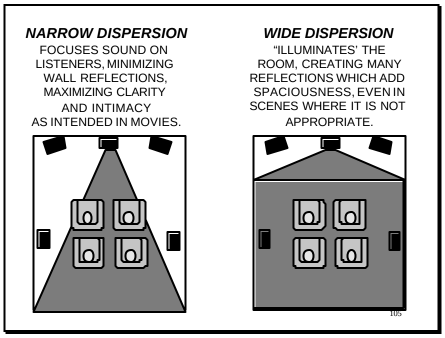
If the wall is too close, the truly fat lady is singing.



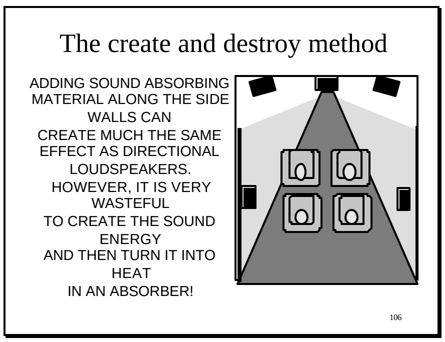
Why? Because the wall is an acoustic mirror, creating a second acoustical loudspeaker, just as it would create a second visual one if the wall were optically reflective. No wonder things got a bit fuzzy.



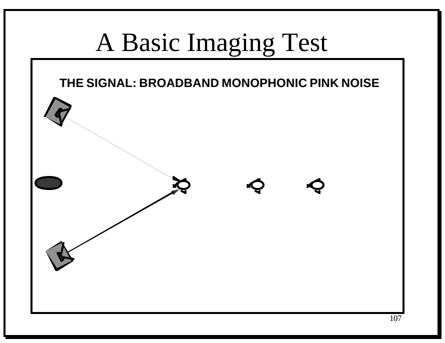
Placing some sound absorbing material on the wall, at the reflection point (have a helper hold a mirror against the wall and find the location where you can see the loudspeaker tweeter from the most important listening position). The reflected sound is attenuated, and the lady loses a bunch of weight.
What material? Acoustic foam or rigid fiberglass board, with or without acoustically transparent fabric covering.
How thick? Not less than one inch, preferably two to four inches.
How large? To be really effective, a patch at least 3 to 4 feet on a side is necessary. Tiny little "cushions" are more psychological than acoustical. Heavy, velour drapes, densely folded also work well.



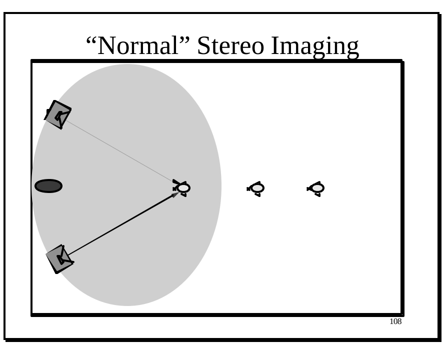
Nowadays, we know enough about horn design to be able to make them sound really good, and take advantage of their directional control. The days of horns that are just loud and sound like megaphones are past – for good engineers at least.
If the room is acoustically live (the way many interior decorators like them), then the only option is to use horns, or waveguides, to control the radiation from the loudspeakers. This way the energy is focused on the listeners, and kept away from the reflecting surfaces, improving the intelligibility and directional effects.



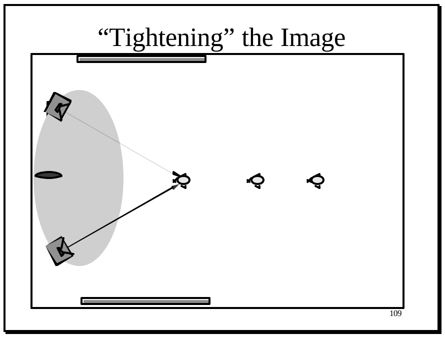
Movies, especially, are designed for listeners in a strong direct sound field. Some people use wide-dispersion loudspeakers, and then cover the walls with sound absorbing material. This gets the job done, but in doing so it makes the entire system work harder, first to create the sound, and then turning it into heat in absorbers. The result, dynamic range is sacrificed. Not necessarily a good tradeoff.
Acoustically dead rooms are also not very pleasant places in which to spend time, conversing or anything else. Some custom home theaters are like this. It is not a recommended solution.



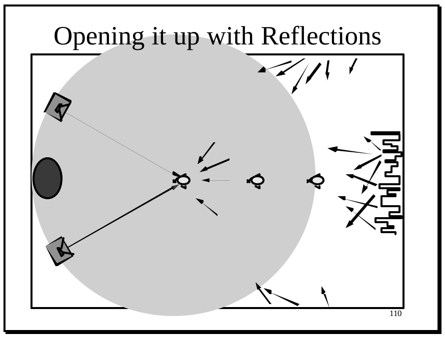
Whether it is a classic two-channel stereo system, or a multichannel system, one of my first tests is to play monophonic pink noise (available on numerous test CD's) through the front left and right speakers, sit in the "sweet spot" and listen. What should be heard is a compact image of noise, floating midway between the loudspeakers. As you move backwards in the room the image should stay. As you lean left or right, the image should move left or right. This is normal. It is a phantom stereo image.



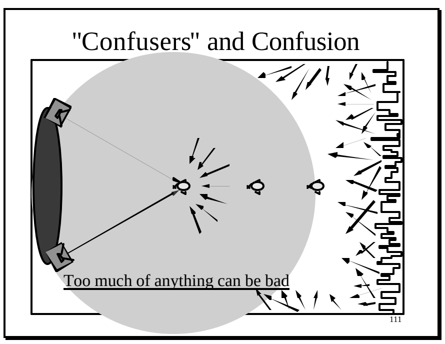
Now, put on some music. The featured artist in pop and jazz recordings should float in the middle location. The band should be across the front creating a solid sound stage (the success of this is greatly dependent on the recording, so be sure to try a few). In recordings with "ambiance", like most in the classical repertoire, you might sense an acoustical spaciousness around you. This is good.



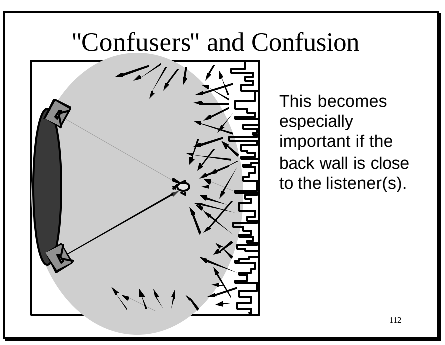
Some two-channel customers like to "get into the image". For them you can suggest some absorbing material, even heavy drapes will do, along the side walls. This attenuates the side wall reflections and the image "tightens" up nicely. Moving the curtains away, opens up the 'space' again.



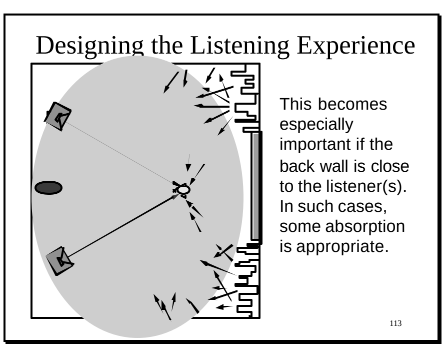
Other customers like to think that they are in the concert hall. For them room reflections are not necessarily a bad thing. In fact, you might consider adding a few more, using some of the commercial diffusing elements on the market. Just be careful not to overdo it. The test is that the center image stays intact even when you move to the rear listening locations.



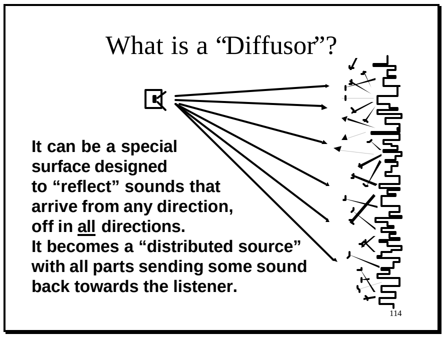
Even good things can be taken too far. I have been in recording control rooms where so much diffusion has been added that the center image is completely destroyed! The noise 'image' was the entire front wall. And recordings were being made in this situation! This design was fashionable – yes there are 'fashions' in acoustics too – a few years ago. Just as in many things, some fashions are just silly. This one was aided by the other fashion of that period: the live-end/dead-end room, another case of an idea taken to excess. It helped some bad studio monitor speakers sound better, but it is not something to be recommended, certainly not for recreational listening, and not for multichannel sound.



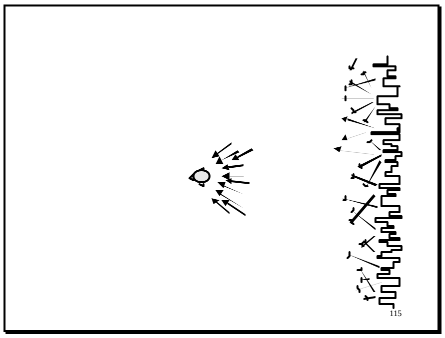
Many homes do not allow us the luxury of sitting away from the back wall. In those cases the last thing one would do is put diffusers directly behind the listeners' heads. Even a hard flat wall can disrupt the front soundstage. A simple demonstration can convince you, or your customer that something is wrong. While listening to the mono pink noise, just hold an upholstered cushion or pillow behind the head of a listener in the stereo seat. Usually the image tightens right up.



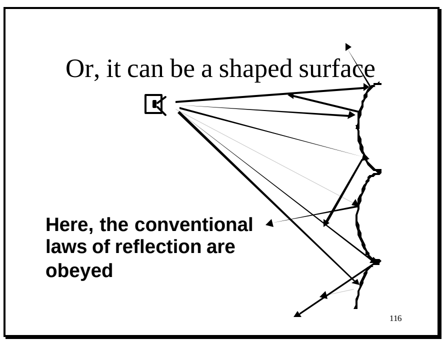
A patch of absorbing material is a much better solution. Use diffusers on the sides, if you like.



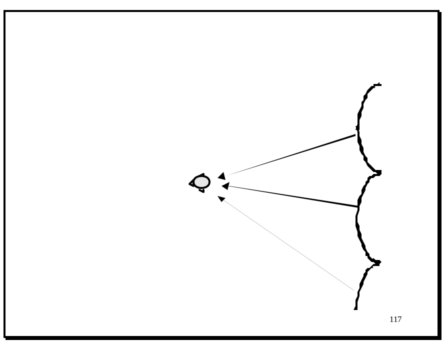
Commercial diffusers are highly specialized devices, designed to accept sounds arriving from any angle, and then to re-radiate them in all directions. Such diffusers, then, need to be considered as distributed sound sources.



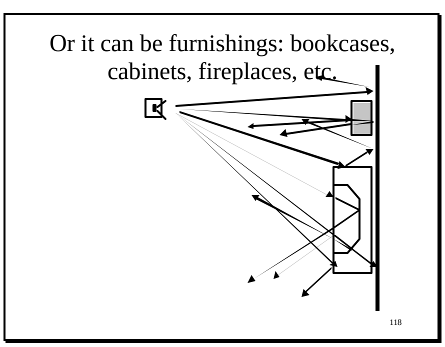
To a listener, these surfaces send a large number of individual reflections to the ears, from all parts of the device.



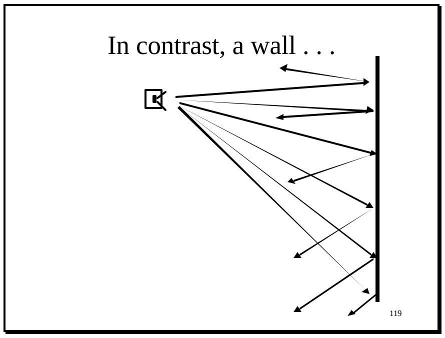
The classic "polycylindrical" diffuser, is nothing more than a curved surface intended to break up large flat surfaces. As diffusers they work very well indeed, and they are inexpensive. They can also be incorporated into interesting looking architectural features, possibly including lighting effects. If you want to get creative, there are many regular and irregular geometrical shapes that work well. A good dry-wall artisan will love you for giving him something interesting to do. Remember to bounce some of the sound vertically too. If the diffusion is to be effective over middle as well as high frequencies, some of the shapes must be a foot or more deep. The notion that textured paint does anything consequential is another fantasy.



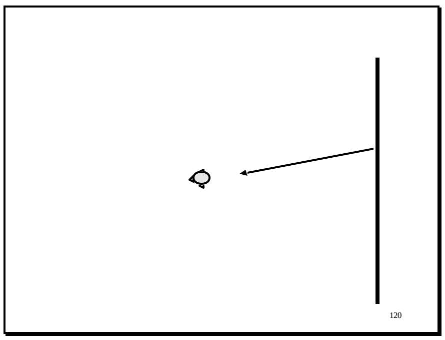
A listener receives only one reflection from each of the curved surfaces.

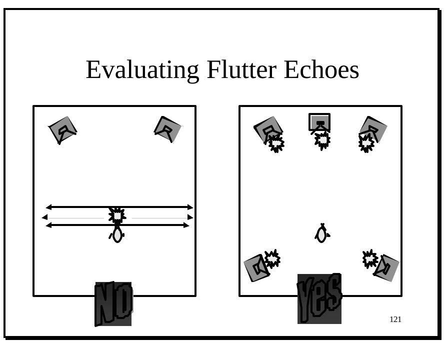


If the listening room is also a normal living space, it may not be necessary to use any special acoustical devices at all. With a little thought, bookcases, display cases, paintings, fireplaces, etc. can all do the job without making the room look at all "technical".

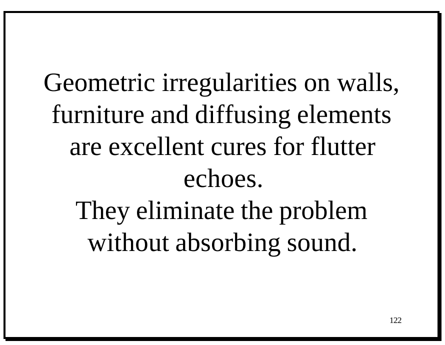


Flat empty walls not only look stark, but they sound that way too.





An acoustical consultant walks into a room, stands by the door, claps his hands, furrows his brow, and pronounces that this room has really bad flutter echoes and you need his (expensive) help to fix it. If this happens, say goodbye.
The only flutter echoes that are important to the quality of reproduced sounds in the room are those that are excited by the loudspeakers themselves. Have an assistant clap hands at the loudspeaker locations while you listen from the relevant locations in the room. If there is a problem then fix it. It matters not that flutters can be heard from the top of a step ladder.



It is amazing how little it takes to cause an audible flutter, and it is amazing how little it takes to get rid of one. I have seen a picture, hung on a slight angle, do the job. Moving a bookcase, adding a wall bulge over a fireplace, a two-foot square patch of diffuser or absorber in a large wall, all have solved annoying problems without absorbing significant sound.

Coming up in Part 3

Perfecting the low frequencies. How many subwoofers? Where do I put them? Where do I sit? What about “bass traps”? How do I get rid of “room boom”?

Here we look at what it needed for truly excellent bass performance in rooms. An understanding of room modes, or resonances, is essential to achieving uniform bass over a listening area. The right kind of equalization can help to make that bass sound good, but it cannot do everything. Some traditional forms of equalization have a good chance of getting it wrong. Interestingly, two or more subwoofers, strategically located, can be very beneficial.

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harman / kardon

Loudspeakers and Rooms for Multichannel Audio Reproduction

by
Floyd E. Toole
Vice President Acoustical Engineering
Harman International

Part 3 – Getting the Bass Right

Choosing the number and locations of subwoofers, and determining where to sit, are fundamental to good bass. Multichannel audio should be shared, so we try to get good bass at several locations. Acoustical knowledge is essential, but EQ can help.

The Rules for Good Sound in Rooms

At middle and high frequencies:
- ✓ Start with a loudspeaker that was designed to function well in a variety of different rooms.
- ✓ Use geometry, reflection, diffusion, and absorption to achieve good imaging and ambiance.

At low frequencies:
- ➢ Maximize the output from the subwoofer(s).
- ➢ Achieve a uniform performance over the listening area.
- ➢ Equalize to achieve good performance.

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So, we think that we know how to pick a good loudspeaker, and with some simple acoustical devices or smart design, we can avoid destructive reflections in a room. Now, what about low frequencies, the ones where the room is the dominant factor?
The first task is to locate the subwoofer where it radiates most effectively. Closed box or reflex (ported) woofers are pressure sources. It matters not which way the diaphragm faces because all such loudspeakers are omni-directional at low frequencies. Obviously, there needs to be 'breathing' space if we choose to face the diaphragm against a wall. And, if there is a port, don't plug it!

Solid Angle VLF Gains

4π steradians	= full sphere = ref. level
	= suspended in free space
2π steradians	= 1/2 sphere = +6dB SPL
	= on floor
π steradians	= 1/4 sphere = +12dB SPL
	= on floor against wall
$\pi/2$ steradians	= 1/8 sphere = +18dB SPL
	= on floor in corner

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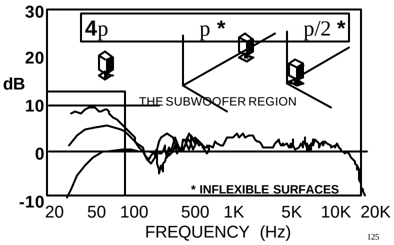
However, where we place the woofer with respect to the adjacent room boundaries does matter. The least effective location is in the middle of a room. It gets better on the floor, still better on the floor against a wall, and best in a corner. This is "best" in the sense of maximizing the quantity of bass radiated into the room.

Measurements in a "Partial" Room



In theory we should be able to get huge gains from corner placement, and the test shown in the slide indicates that, if conditions are right, the gains are there. In real houses, the gains are a lot less, because of flexible room boundaries, open archways, etc. Still, even a 3 dB gain doubles the acoustical power into the room, and that is equivalent to adding a second woofer. Not bad, and it's free.

A Subwoofer in a Corner

- Produces the maximum possible LF output.
  - This is good
- Energizes all of the room modes
  - This can be good or bad, depending on the room, and how it is arranged.

CONCLUSION: Start with the sub in a corner, and move it only if necessary.

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However, as they say, there is no free lunch. Corner locations excite every resonance in the room. This may or may not be bad. You will learn how to know the difference before we are through. If it is good for a particular room, then leave it alone and move on to the next problem. If not, find a better location. However, a corner is a logical starting location. For many simple installations, an unequalized single woofer in a corner can work very well indeed. In fact, I am listening right now to just such a system in my office/den.

The Rules for Good Sound in Rooms

At middle and high frequencies:

- ✓ Start with a loudspeaker that was designed to function well in a variety of different rooms.
- ✓ Use geometry, reflection, diffusion, and absorption to achieve good imaging and ambiance.

At low frequencies:

- ✓ Maximize the output from the subwoofer(s).
- ➢ Achieve a uniform performance over the listening area.
- ➢ Equalize to achieve good performance.

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A multichannel audio system is intended to be a social experience, shared among several listeners. It is not like stereo, an antisocial experience, if it is to be heard properly.

We need to try to create a situation in which as many listeners as possible hear essentially the same sound.

The problem here is Standing Waves, Room Resonances, Room Modes, Eigentones, etc.

These are all the same phenomenon.

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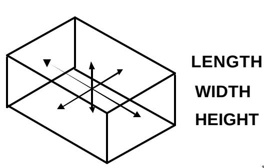
Most rooms "boom". They have their own bass personalities. It is simply not possible to listen to the true bass output from a loudspeaker unless it is done in an anechoic space, and at very low frequencies, that means outdoors. What we hear indoors, also includes the room, and the positional factors within it.

Classes of Room Modes

- AXIAL: occurring between opposite parallel surfaces



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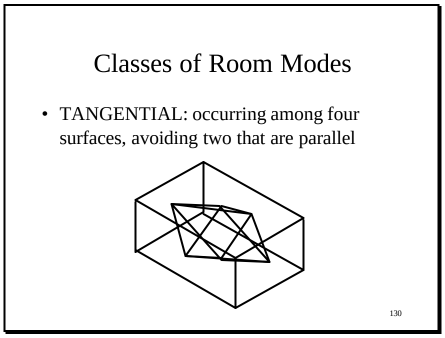
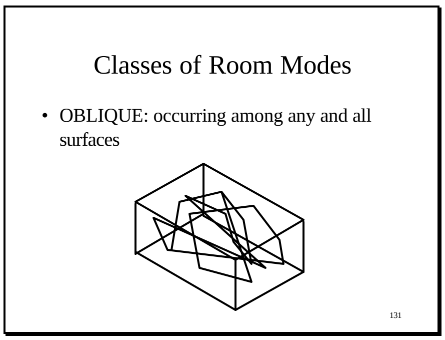
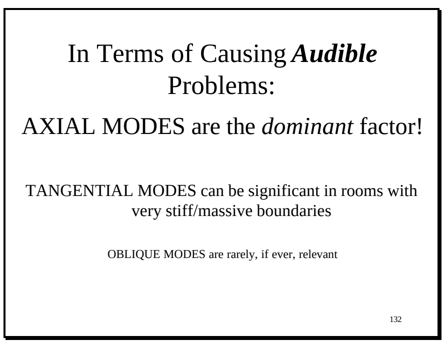
Sound reflects back and forth between two parallel surfaces. At certain frequencies the incident and reflected sounds conspire to form "standing waves" in which those frequencies can be amplified, and what we hear at those frequencies depends on where we and the speakers are located. These are axial modes, since they exist along the major axes of the room.

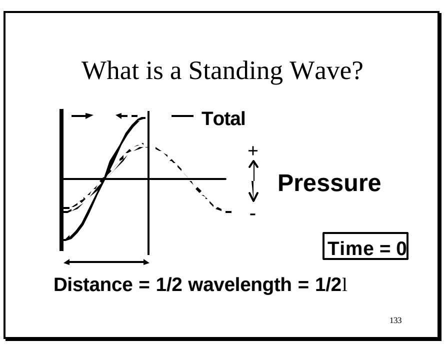


The cyclical reflected patterns can include two, four or more surfaces. If four are involved, we call them tangential modes. Since some sound is absorbed at each reflection, generally the tangential modes are less powerful than the axial modes, which include only two reflections per cycle.

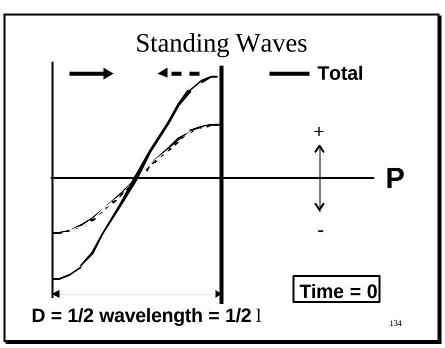


Oblique modes can involve any and all surfaces. They tend, therefore, to be the least important.





The simplest of the standing waves, the axial modes, exist between two parallel reflecting surfaces. In this slide shown here, imagine that the two vertical lines represent walls in a room. Imagine a woofer against the left-hand one, radiating a pure tone – a sine wave – as shown. If the right wall were not present, the sine wave would simply propagate away. With the wall in place, the portion of the sine wave that would have moved on to the right is reflected back towards the source. The walls here are separated by exactly one-half wavelength, so the reflected sound wave is identical to the incident wave – they overlap perfectly. When added they make the "total" sound wave shown. Remember, this is a "stop action" view.



Here is the same instant in time expanded in scale.

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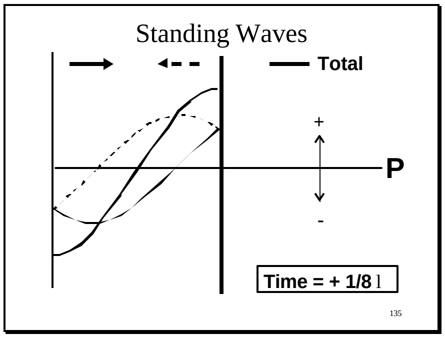
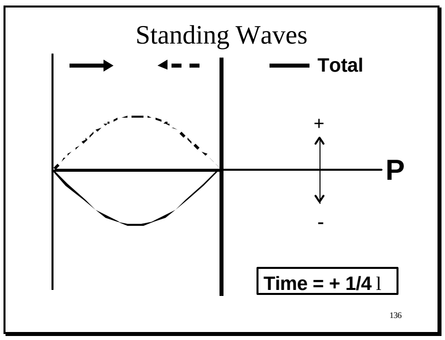
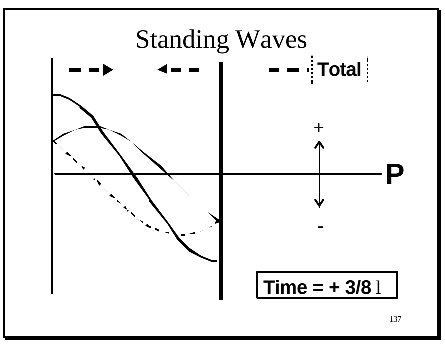


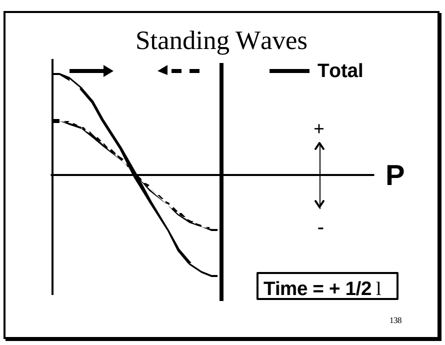
Here, we have done a very, very quick "play" and "pause", stopping the action one-eighth of a wavelength later. Now the incident and reflected waves have different shapes, but the "total" is much the same, only a bit lower in amplitude. Interestingly it still crosses zero pressure half way between the walls.



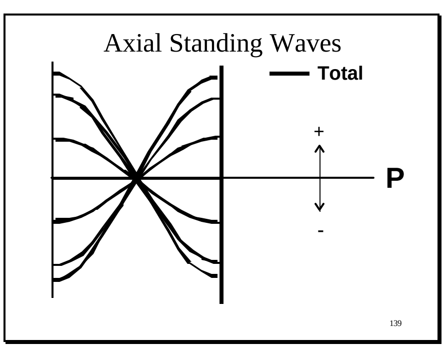
This is another 1/8 wavelength later, or ¼ wavelength from where we started. The instantaneous sound pressure is zero everywhere.



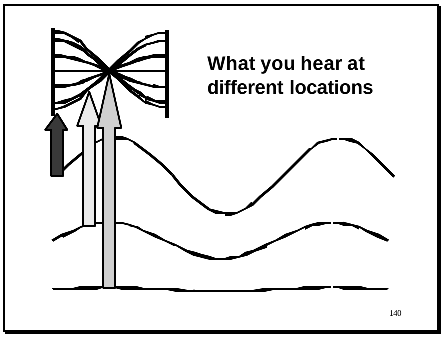
Moving on yet another 1/8 wavelength, we note that the total waveform has "flipped" polarity, like a see-saw pivoting at the half-way point across the room. Still no sound at the mid point.



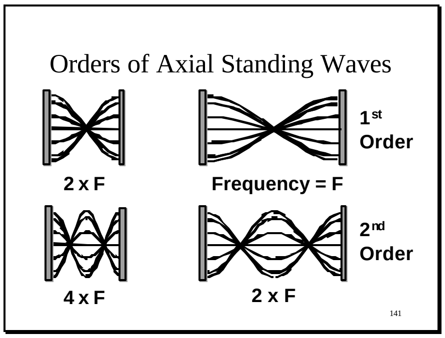
At the half-wave stop-action view, we are precisely in a situation that mirrors where we began. If we continued for the remaining half wave, we would end up where we started.
So, for the frequency at which the room dimension is precisely one-half wavelength, there will be a resonance, and a standing wave in which the sound pressure is always zero at the mid point between the reflecting surfaces. On either side the sound gets louder as one moves towards the walls. Note, that the instantaneous polarity of the pressure change is opposite on opposite sides of the pressure minimum, or null. This is important. Remember it.



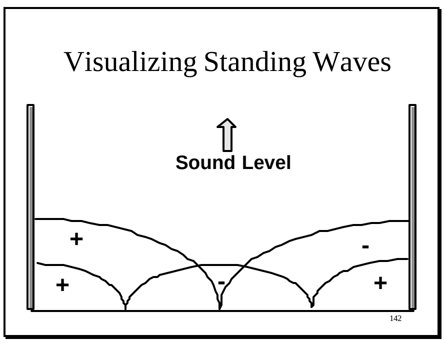
Here we see a superimposition of the stop-action "total" waveforms for one complete wavelength.



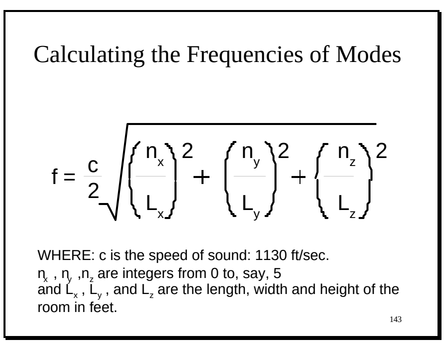
There is minimal sound at the mid point, and the sound gets louder as we move towards the walls.



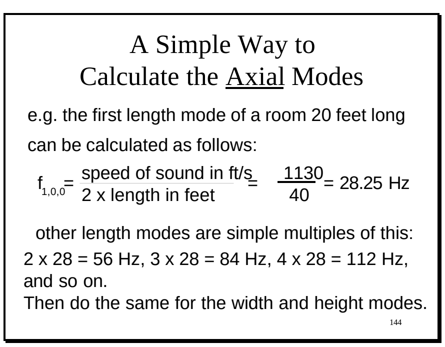
For a given distance between reflecting surfaces, the first resonance will be at the frequency having a wavelength equal to twice the separation. If the distance is halved, the frequency is doubled. There is also a second-order resonance, or standing wave, at exactly double the first frequency, and this standing wave has two nulls. There will also be third, fourth, and so on.



To visualize the standing waves, let us just plot the sound pressure as a function of distance, and remember that the polarity changes each time we cross a null.



This formula allows us to calculate all of the possible resonances in a rectangular room. It is a bit frightening.



For the majority of situations, it may be sufficient to calculate only the axial modes. If so, it is very simple. Measure the room dimensions. Multiply the length by two, and divide it into 1130 (for dimensions in feet) or 345 (for dimensions in meters). This gives this first-order resonance along that dimension. Multiply that frequency by 2 for the second-order resonance, by 3 for the third-order, and so on. Usually it is necessary only to look at the first three or four orders. Now, repeat that for each of the other two dimensions.

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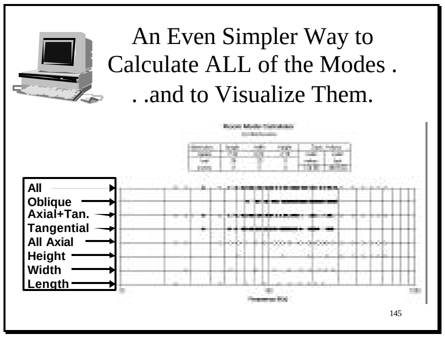

If you have a computer, it is all even easier. Just type in the dimensions, and the work is done for you. This is a little spreadsheet program that runs in Microsoft Excel for PC's that is available for download from www.harman.com in the section "white papers". It shows, in an easy to understand form, all of the modes in a rectangular room.

It has long been believed that, to be good, a room must have a uniform distribution of modes in the frequency domain.
This is certainly true for reverberation chambers, where the notion originated. But, is it true for listening spaces???

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But what about the "ideal" room?

The distribution of modes in the frequency domain is determined by the RATIO of the room dimensions:
Height:Width:Length
e.g. 1:1.5:2 = 8 x 12 x 16 feet

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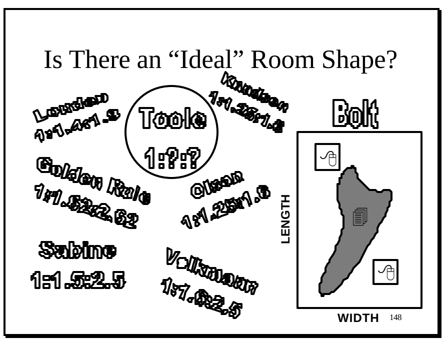
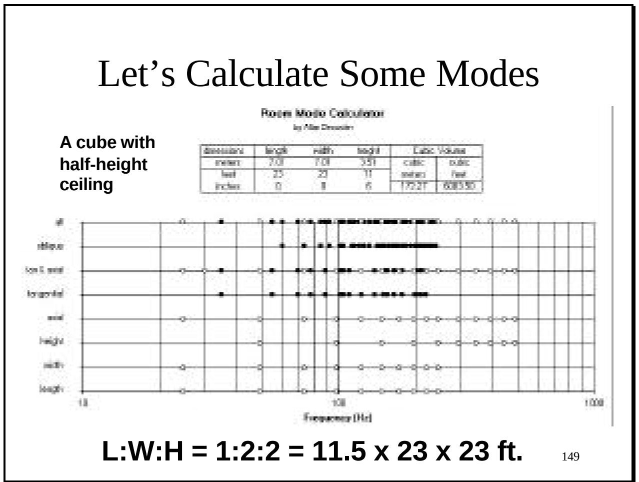
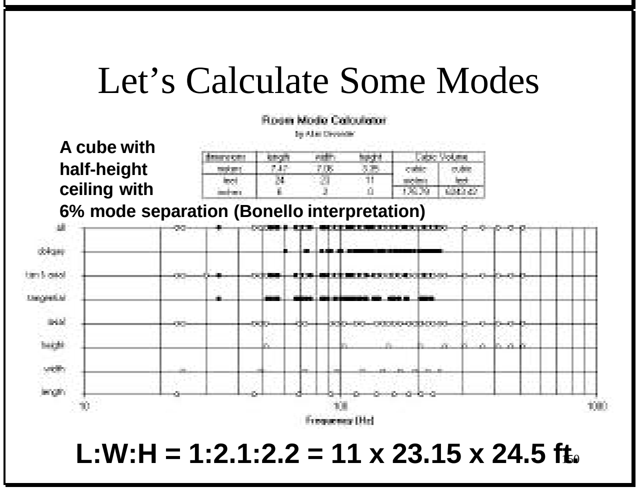


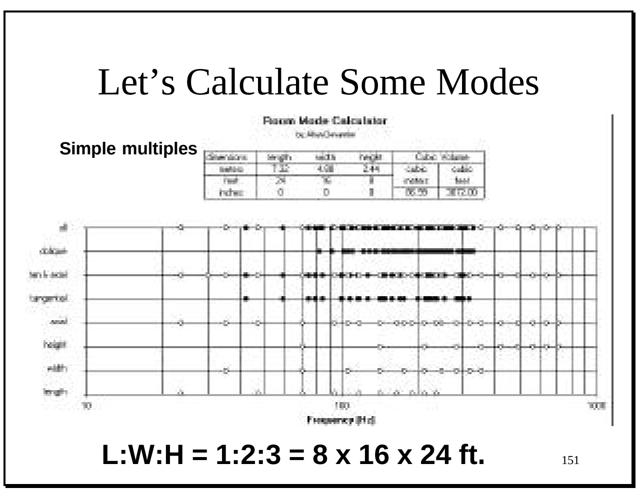
Over the years several acoustical luminaries have lent their names to room dimensions that purport to have advantages. Do I have a favorite too?



Everyone knows that a cube is the worst possible shape. Such a room is really impractical in any event, because the ceiling would be very high. Let's compromise, and make the ceiling height half of the other two dimensions. It can be seen that the room resonant frequencies all line up like little soldiers on parade, with big gaps between them. The suggestion is that some frequencies will be overly accentuated, and others not adequately represented. A room should be better if it had a more uniform distribution of resonant frequencies.



It has been suggested that adjusting the room dimensions to produce a slight separation of the resonant frequencies should help. Here, a recommendation of 6% separation of frequencies has been followed, and it is seen that while the high frequencies appear to be improved, the low frequency resonances are still in closely-spaced groups with large gaps.



A room with dimensions that are simple multiples of each other produces a mixed situation. Some of the modes seem to be well separated, but others line up at certain frequencies.

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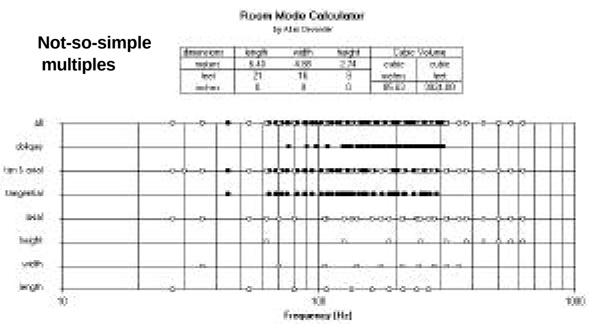
Let's Calculate Some Modes

Not-so-simple multiples



L:W:H = 1:1.8:2.3 = 9 x 16 x 21 ft. 152

Just departing from simple multiples seems to do wonders. Here the modes all seem to be quite well distributed. If this approach to room design has any merit, this room should have some audible advantages. Does it?

This all makes a very nice story, but does it <u>really</u> matter?

Maybe . . . Somewhat . . . It all depends . . .

Oh, all right,
No!

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Why not?

- The calculations assume that all of the modes are equally energized by the loudspeakers –they are not.
- The calculations assume that all of the modes are equally heard by the listener(s) – they are not.
- The only modes that matter, are those that are involved in the transfer of sound energy from the loudspeakers to the listener(s).

So, how do we determine that?

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The simplifying assumptions underlying these predictions make them simpler, but also simply invalidate them!

Important Facts About Woofers, Listeners and Standing Waves.

- Conventional woofers are *sound pressure* generators. They will "couple" to the room modes when they are located in *high pressure regions* of the standing waves.
- Ears respond to *sound pressure*, therefore, room modes will be most audible when our heads are located in the *high pressure regions* of the standing waves.

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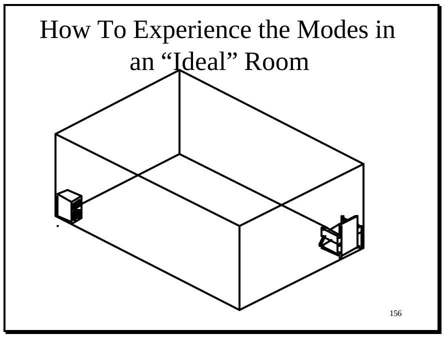
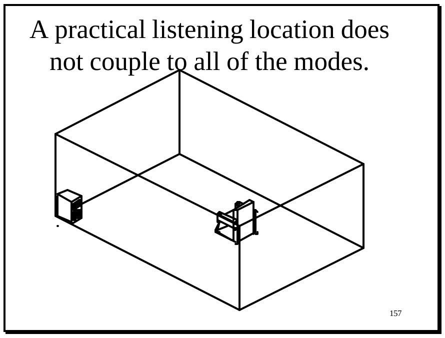
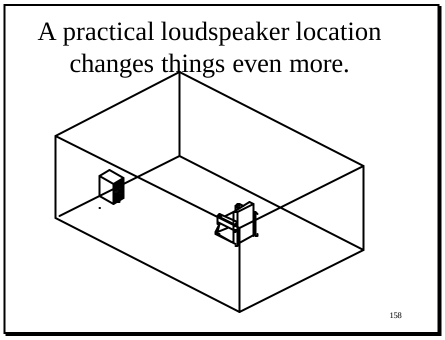


In order to hear the benefits of a room with "ideal" proportions, this is how it would need to be arranged. The ideal proportions were determined by looking at the distribution of all the resonances, so we need to energize them all. A woofer on the floor in one corner excites all of the room resonances. Likewise, the listener must sit with his head stuck in another corner, in order to hear all of the modes.

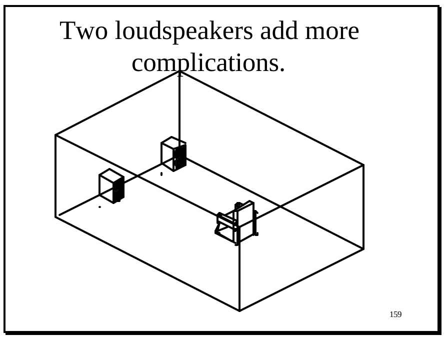
This sounds silly, but it is true.



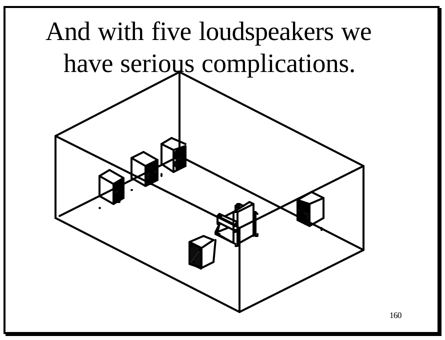
Of course, in practice, we don't do this. We sit where we want to.



And we place our loudspeakers where they need to go for good imaging.



And we listen in stereo.



If not in a multichannel format. All of those neat calculations don't mean a thing in a situation like this! They are just an academic exercise, more window dressing.

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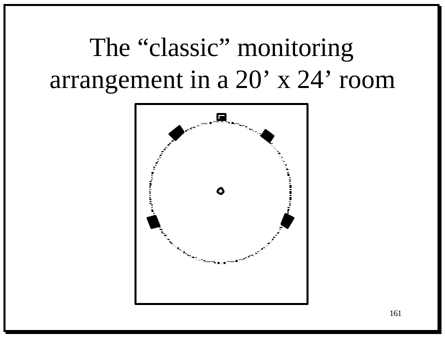
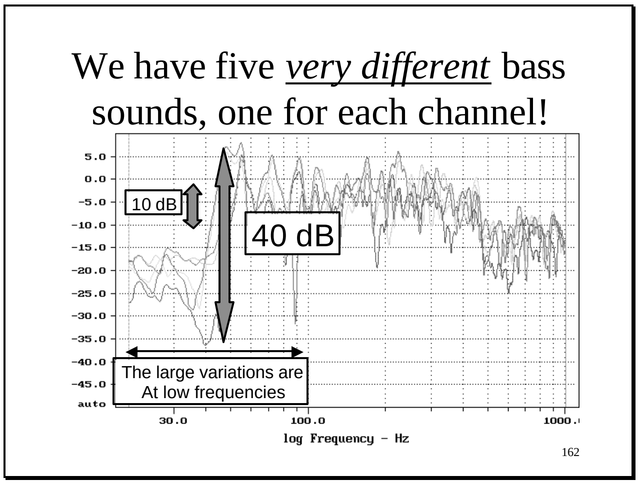
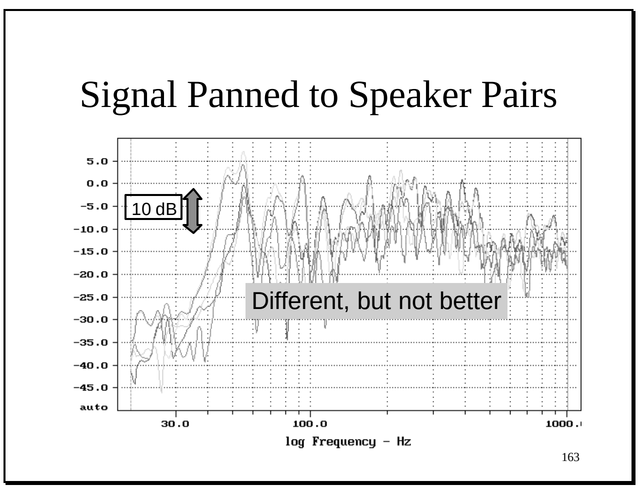


Here is how some professionals listen, with five full range loudspeakers located according to the new European standard: center, +/- 30° and +/- 110°.

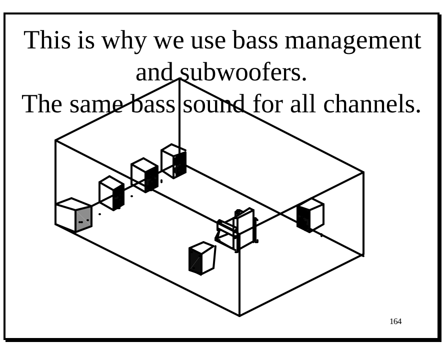


When a full-range signal is panned to each of the loudspeakers in turn, and measurements are made at the listening position, we find hugely different bass responses for each of the loudspeakers. The differences are a large as 40dB in this room, and the biggest ones are all at low frequencies. The reason, the woofers each have very different acoustical “coupling” to the room resonances because they are in different locations. This will be different for every different room. Again, referring back to the “circle of confusion” the bass that was heard in the control room will not be the same as that heard at home. It cannot be.



Attempting to improve the situation by panning the bass to pairs of loudspeakers changes things, but does not remove the problem.

Anybody think that an “ideal” room can help this? An anechoic room would, but none of us would wish to live in one.



And this is why bass management and subwoofers make sense. Now we can place the woofers where they perform optimally for a specific room with a specific listening position. We can place the satellites (a term that seems inappropriate for some of the large capable loudspeakers that we use in the high-passed channels) where they need to be for directional and imaging effects.
In other words, we design the low-frequency portion of the system separately because rooms force us to do so. This is the only way that we can get good bass in any room, and have any hope of having similarly good bass in different rooms. Remember about preserving the art?

How many subwoofers?
Where do we put them?

Where do we sit?

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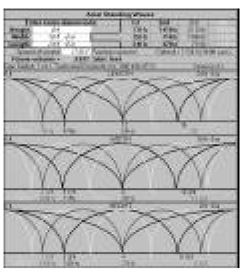
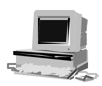
More than one subwoofer may be needed for high sound levels in large listening spaces. It's physics. One huge box might be difficult to hide, while a number of smaller ones might be less conspicuous. But there are other considerations which become obvious as soon as we start looking in detail at room resonances.

'Fast' and 'Slow' Woofers.

A digression here about the "speed" of woofers. I keep on hearing stories that small woofers are "faster" than big ones. Well, there is truth in the argument if you consider the highest frequencies they are capable of reproducing. However, if we are crossing them over at 80 Hz, for example, to use them as subwoofers, we have limited that highest frequency to be the same for all. They are then all equally "fast". Woofers are minimum-phase devices (see Part 2). Their time-domain behavior – speed, punch, drive, pace and rhythm - can be anticipated from their frequency responses. We will soon see that rooms really mess this up and determine what we hear.
As for moving mass, we use bigger motors on larger, heavier, diaphragms. It's a "horsepower thing".





A Standing Wave Calculator

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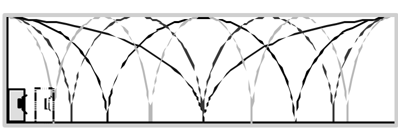
Understanding what happens in rooms at low frequencies requires knowledge of what the standing waves, or resonances, are doing to us. Part of the Excel program mentioned earlier is a great help. It shows the pressure distributions for the first few axial modes along each of the room dimensions. It is available for download from www.harman.com under "white papers".

Woofer Location Decides How Much Energy Each Mode Receives.



A woofer on the floor, against a wall, energizes all modes along that axis.
What happens if we move it forward, say, to the first null?

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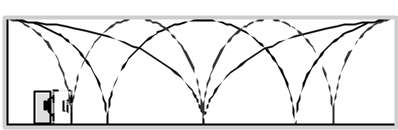
Here I show a woofer placed against a wall, where it excites all of the modes along the length of the room. Why? Because it is in a high-pressure region for all of the modes. What happens if we move the woofer forward, to the location of the first pressure minimum?

Woofer Location Decides How Much Energy Each Mode Receives.



A pressure source located at a pressure null, or minimum, does not energize that mode.
What happens if we move it forward to the second null?

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That particular mode is not energized by the woofer, and it disappears. What then happens if it is moved further ahead to the next null?

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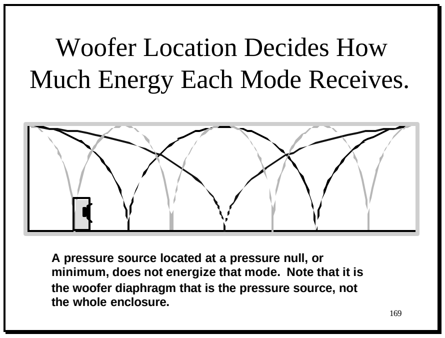
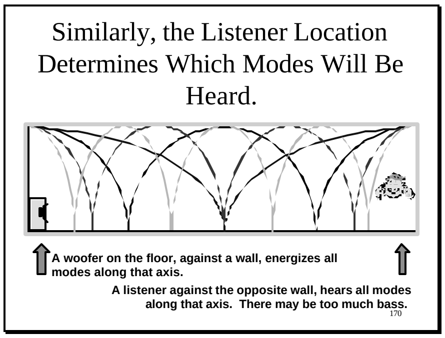
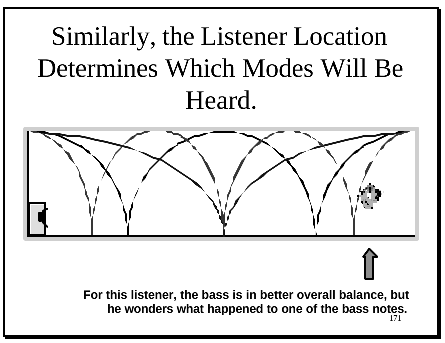


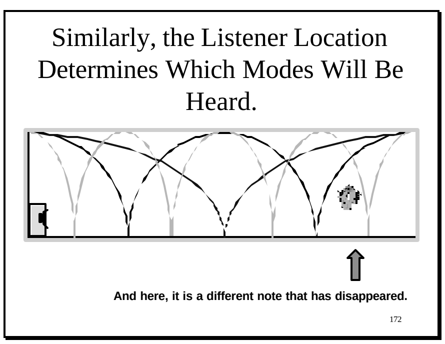
That mode ceases to be activated, but the other one returns. Woofer location determines which of the room resonances is activated, and which is not.



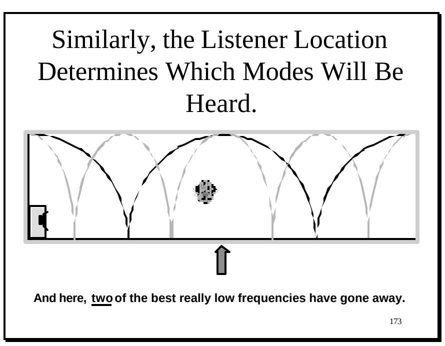
Just as a woofer against a wall energizes all of the standing waves in that dimension, a listener with his head close to the opposite wall hears all of the modes. Just as a woofer against a wall enjoys a gain in output because of the adjacent boundary, so the ears enjoy a similar low-frequency gain. There will be too much bass!!



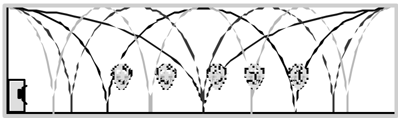
Moving the listener forward provokes the same problem we had when moving the loudspeaker forward. If the head is at a null, no sound will be heard from that mode.



Different positions mean that different frequencies will be heard with sometimes very different loudness.



Similarly, the Listener Location Determines Which Modes Will Be Heard.



But, there are several locations where happiness might be possible.

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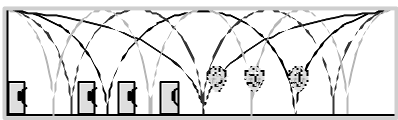
However, there are locations that might work reasonably well.

Similarly, the Listener Location Determines Which Modes Will Be Heard.



These are also suitable locations for loudspeakers, but not all are practical, just as not all listener locations are practical.

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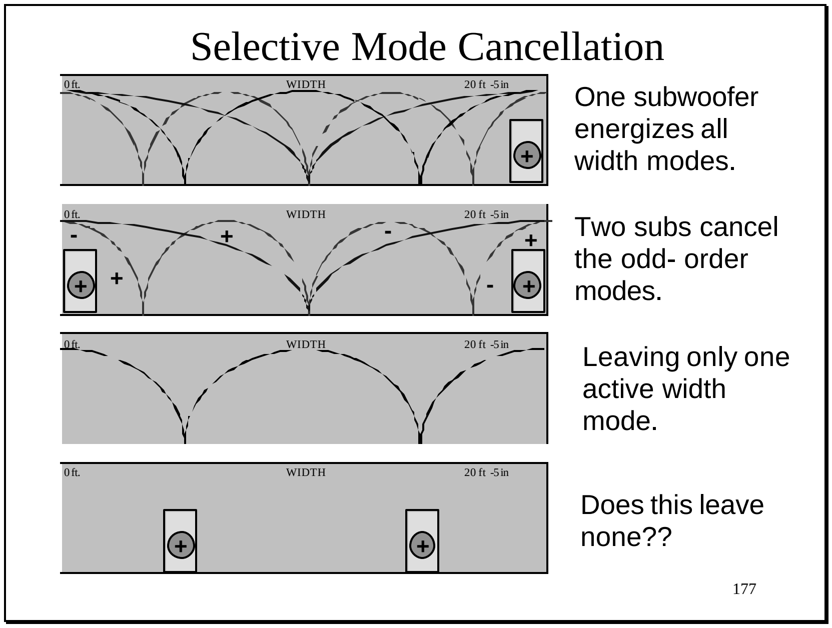
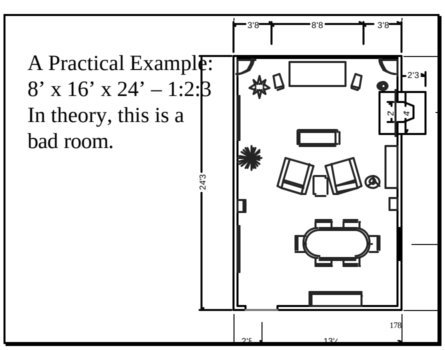
And this is true for woofers as well as listeners.



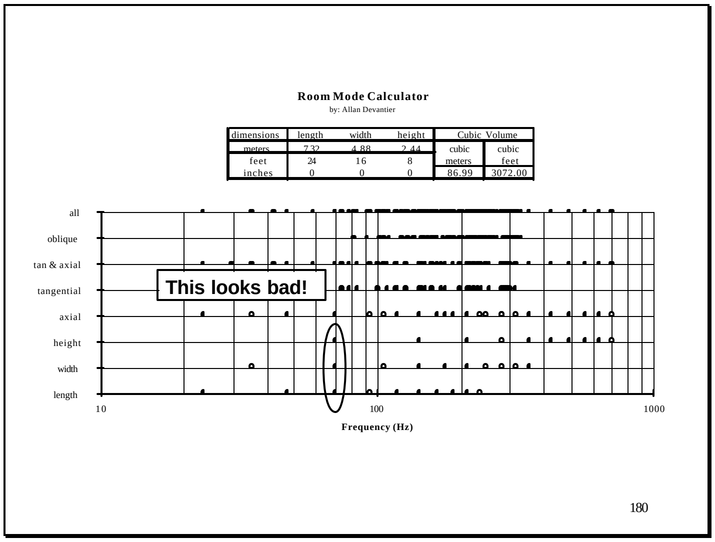
Here we get a bit fancy. Remember the standing wave diagrams showing the "see saw" behavior of the waveform pivoting on a null. It means that at any instant in time, the sound pressures on opposite sides of a null in a standing wave are in opposite polarity. If one side is increasing, the other will be decreasing. None of this matters if we have only one source of low frequency energy in a room. However, if we have two or more, things get complicated. The top picture shows all of the width modes in a room being excited by a single subwoofer. If we place another in a symmetrical position on the other side of the room, the woofers, which receive exactly the same signal, are operating in phase (the same polarity) with each other. The first- and third- order modes, however, exhibit opposite polarities at the subwoofer locations – see the opposite polarity signs. What happens? The subwoofers couple in a destructive manner with the odd-order modes, and those modes are simply not energized. They do not exist. This leaves only one mode across the width of the room. If we get even more clever, and move the subwoofers to the null locations for that mode. They are still in opposite polarity regions for the odd-order modes, and the result is that all of the width modes are significantly attenuated, if not eliminated. Magic, no. Science, yes.

So, why would we want to do this? Aha! Would it not be a good idea for everybody in each row of a home theater to hear the same bass sounds? Would it not be a good idea for a recording engineer to be able to move from one end of the console to the other without experiencing huge changes in bass? Well, this is how it can be accomplished. We are not saying, yet, that it is good sound, merely that it is the same sound. Once things are equalized in the sense of getting everybody hearing more or less the same sound, we then may need to equalize in the sense of changing the frequency response of the system.

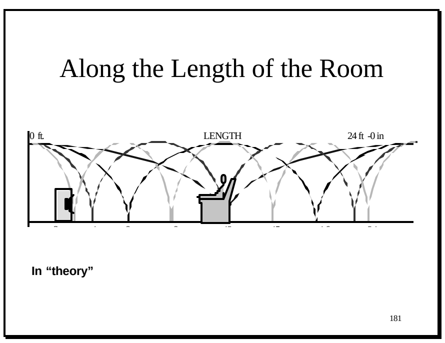


Just to prove a point, let us take a real example: a room that, by most advice should be a bad room. It is not an unusual situation. There is a living/dining room, with a rear projection television at one end, seats in the middle area, and a dining area behind the chairs.

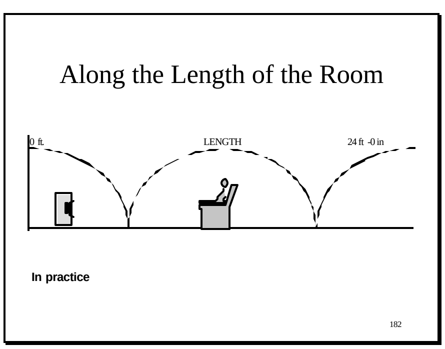
It looks normal, and could happen in a detached dwelling or in an apartment.



The modal distribution shows that there could be a problem with three modes stacking up at 70 Hz.



Fortunately, we have some help from the locations of loudspeakers and listeners. The loudspeakers are lined up with the front of the RPTV, and that puts them in the null of the fourth-order mode. The listeners are seated at the mid point, and they are in the nulls of the first- and third-order modes.



That leaves only the second-order length mode to cause problems.

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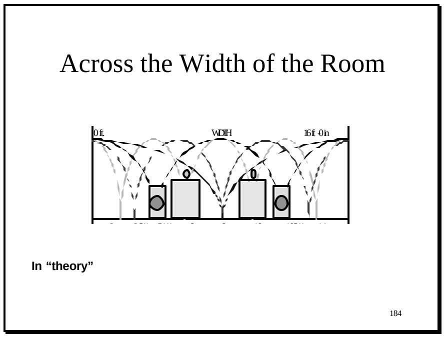
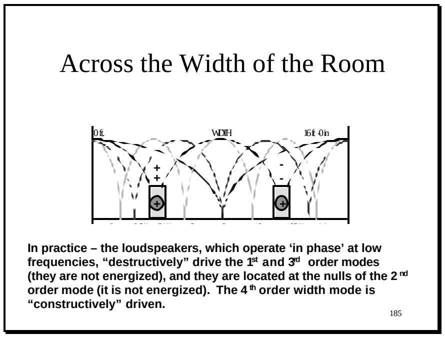
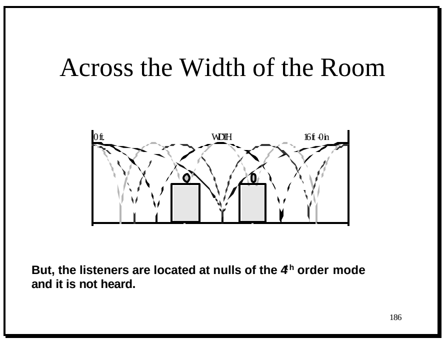
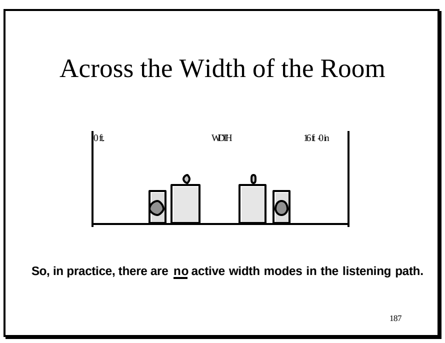


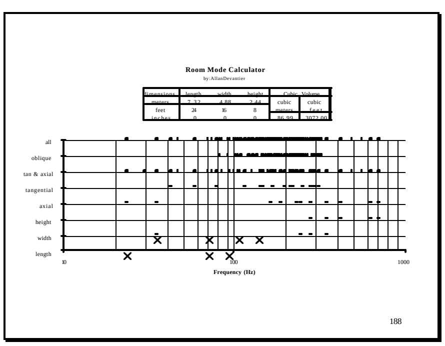
In theory there are a bunch of modes at play.



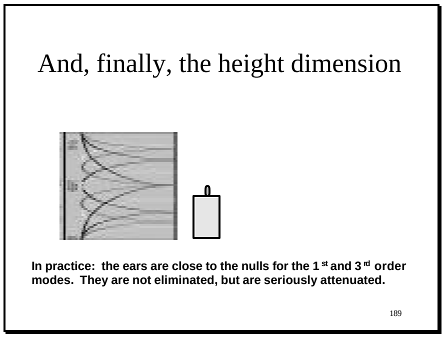
In practice, we have selected locations for the loudspeakers and listeners that avoid them all.

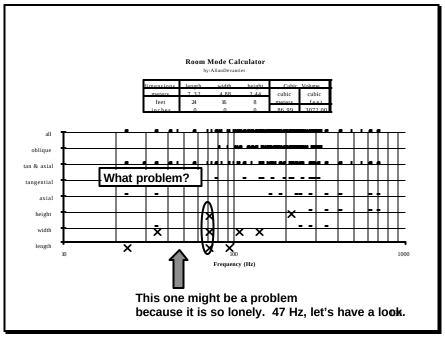






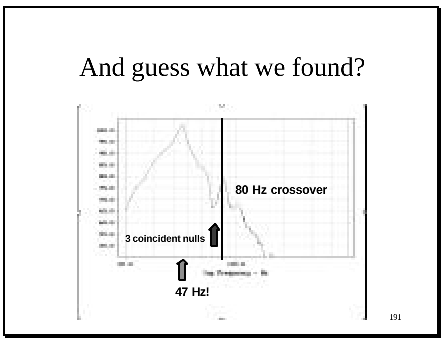
The modal distribution looks a lot less cluttered after we cross off all of the modes that are not involved with communicating between the woofers and the listeners in those chairs. Remember, things will be different elsewhere in the room.



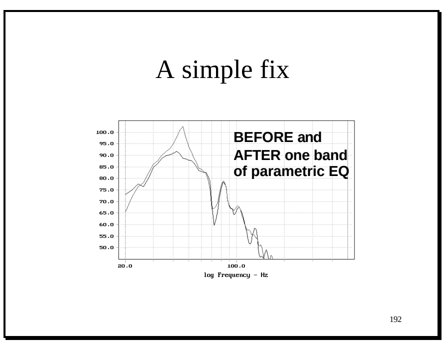


When we are through, there is only one solitary room resonance that is actively involved in the acoustical link from the loudspeakers to the listeners. Amazing!

Let's measure it and see what we have.

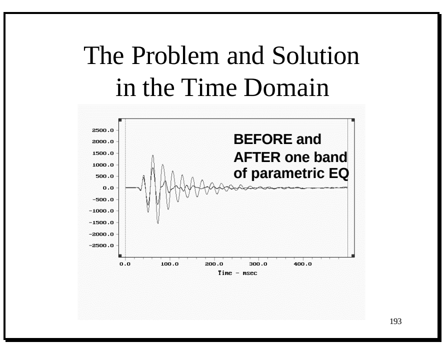


Surprise, surprise. There is a resonance at 47 Hz, the frequency of the remaining second-order length mode. And, guess what, there is a sharp dip at 70 Hz, just where we successfully canceled those problematic modes. To get rid of the dip, all we need to do is to be less successful at canceling some of those modes. In other words, move a chair or a loudspeaker a few inches.



And, what about the resonant peak? When we listened, it was clearly audible, making kick drums "boom", and all bass inarticulate and floppy. All bass tended to be "one note bass", the "tunes" in the bass guitar were all but gone. We could have blamed the woofers, accusing them of being slow, uncontrolled, and tuneless. But we know better.

The solution? Because the room resonance is a minimum-phase phenomenon, we designed a parametric filter to attenuate the peak, and the system is instantly transformed. The bass was tight, the guitars played tunes at low frequencies, explosions no longer had a pitch. The room sounded great.



The long ringing of the original room is transformed into a well damped, tight, transient response. It works. The woofers were 'fast' all along. We just couldn't hear it.

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So, according to dimensional ratio theory, this was supposed to be a bad room. In this practical example, it ended up having only **one** problem resonance and, after equalization, it yielded truly superb sound!

Good Sound in a "Bad" Room!

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O.K. So this is showing off. However, it does prove a point. If you understand basic room acoustics, have some decent analytical tools at your disposal, including measurements, most rooms can be under control.

Not all stories have such a happy ending, but happy endings are unlikely unless the loudspeaker and listener locations are evaluated in conjunction with the standing wave patterns.

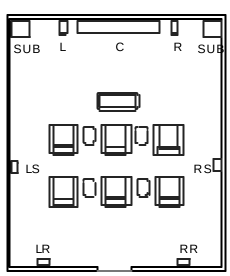
195

REMEMBER:
The only modes that matter are those that participate in the communication of sound from the loudspeaker to the listener!
and
There is NO 'ideal' room!

196

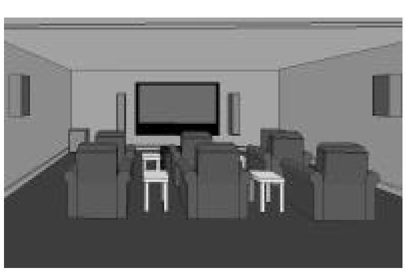
So, back to the issue of "ideal" room dimensional ratios. Personally I have no favorites. I have yet to encounter a room that could not be made to sound at least good, if not excellent.

So, what happens if we start with a room that is "good" to begin with?

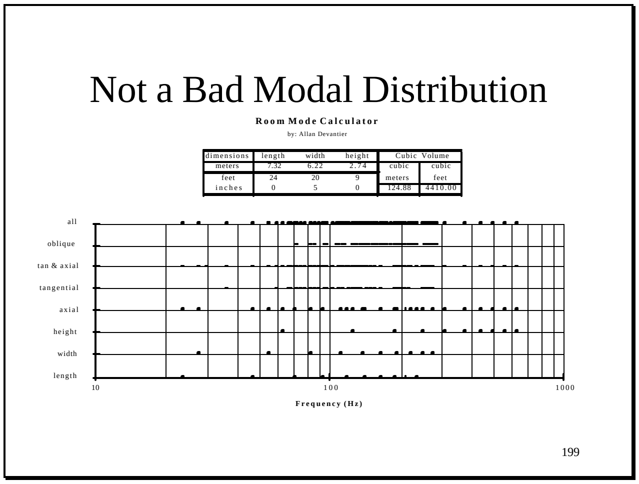


197

In this example we will really make use of acoustical measurements. It will be seen that high-resolution (better than 1/3 octave) measurements allow us to confirm which resonances are active in which parts of the room. They will allow us to experiment, intelligently, with subwoofer and listener locations. In short, good measurements put us in control. We may or may not achieve our fantasy performance, but we certainly can get closer than would be possible by trial and error.

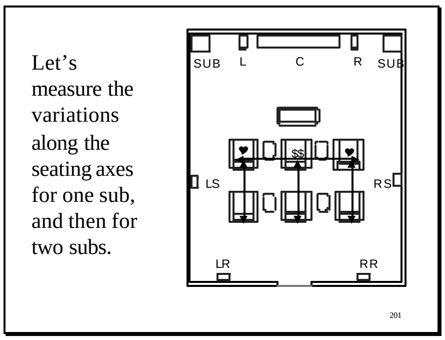


198

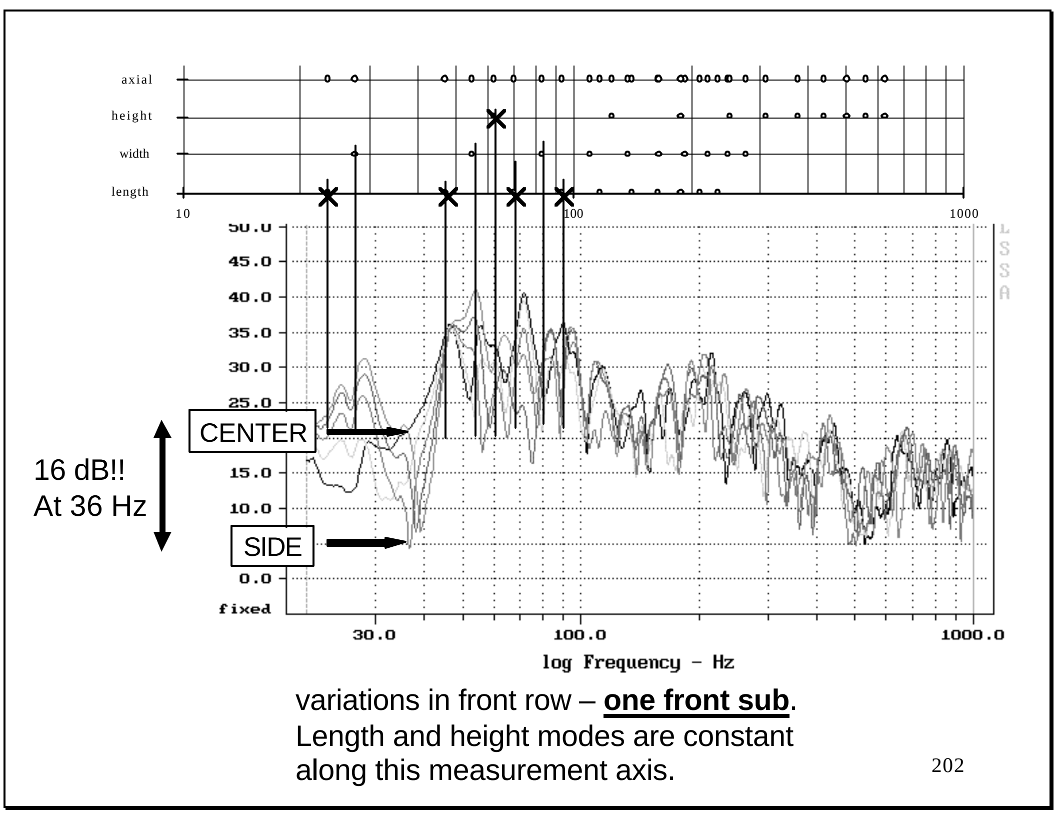


According to common belief, this room should have some advantages, because of the favorable distribution of modes.

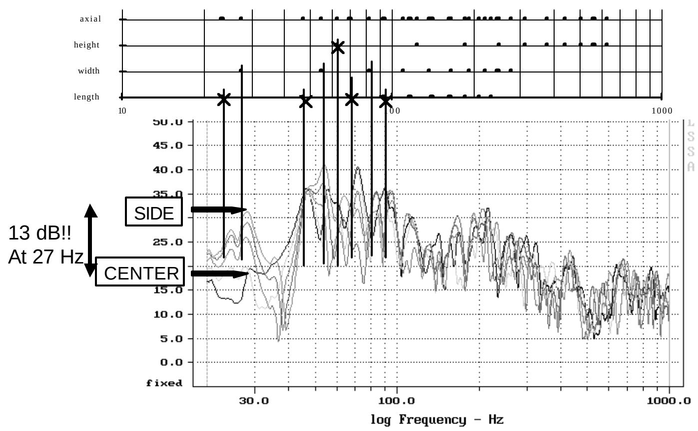
However, taking nothing for granted, let's actually measure what is happening at different listening locations, when we employ one or two subwoofers.



It is obviously important to keep the customer and the family or close friends happy, so pay attention to the front row. I think it is natural to expect the best of everything in the front row.
Measurements are made at 18-inch intervals across the front row, and from each of the seats to the rear.
Note, first of all, in the slide below, that it is possible to identify many of the axial modes calculated for this room. The frequencies don't always line up exactly, and this will be explained later.



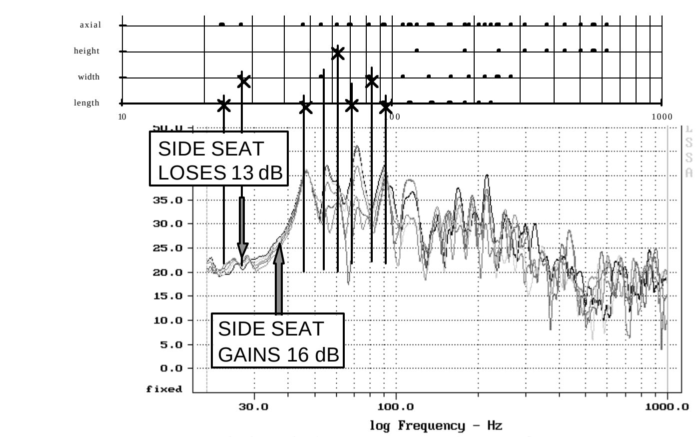
The above graph shows, in detail, what happens at 18-inch intervals across the front row. Because the room is symmetrical, what we are really looking at is what happens from the center seat to the side seats. It can be seen that there is a 16 dB difference around 36 Hz between the center and the side seats. This is huge, and this is a very important part of the frequency range. It could be that the host, sitting in the center, exclaims to his buddy, to whom he is showing off the system, “listen to that fantastic bass!”. The buddy, sitting beside him, could be justifiably underwhelmed if the bass frequencies fall into this range.



variations in front row – **one front sub**.
Length and height modes are constant along this measurement axis.

203

Later in the same movie, there is a thunder storm, and the almost-subsonic rumble is perceived by the guest to be just stunning, but the host thinks it is just O.K.

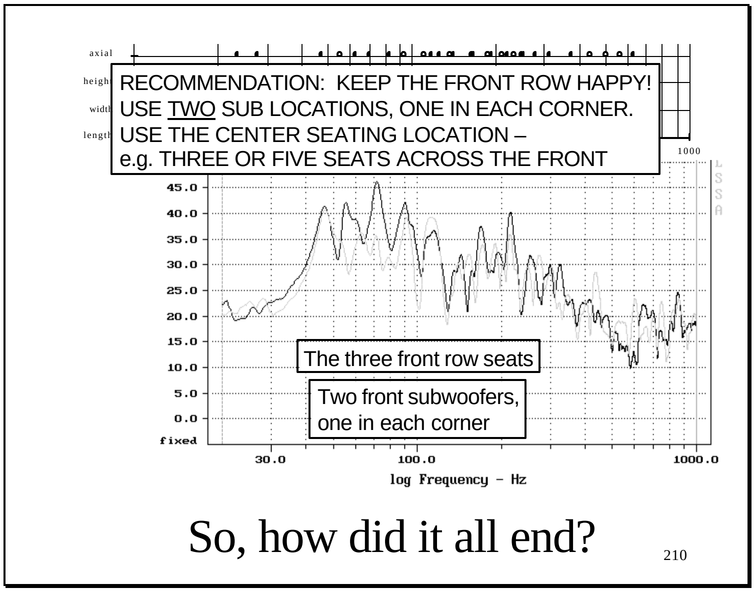


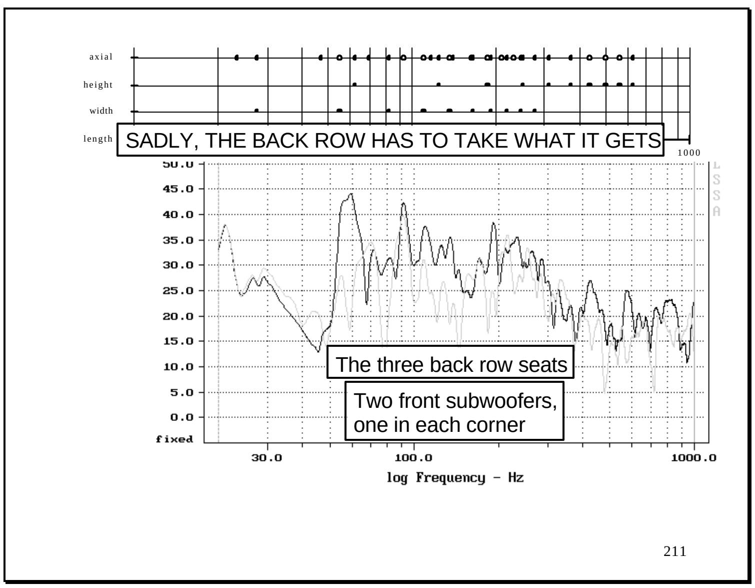
variations in front row – **two front subs**.
The odd-order width modes are now cancelled.

204

This is where mode canceling can be useful. Using a pair of subwoofers, the problematic width modes are cancelled, and the bass is rendered uniform across the front row up to about 55 Hz. There is a trade-off. We sacrifice some of the deepest sounds in the side seats, but we pick up 16 dB of the more important frequencies higher up, around 36 Hz. The recommendation is, therefore, to use two subwoofers located opposite each other across the front of the room. The bass response is still uneven, but we have managed to make it more uniform across the all-important front row of seats.

The pair of subs will not only deal with the front row, but will also make performance across the back row more uniform. However, as it is set up, the front and back rows will hear different bass sounds. A decision must be made of how to equalize, if that route is chosen. Normally, priority is given to the front row

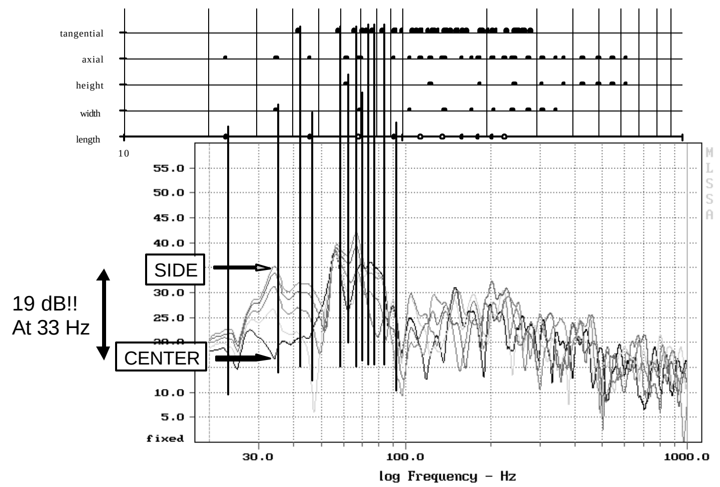




Now, let us look at a smaller
room – 24 x 16 x 9,
with 2” x 6” studs,
and two layers of
5/8” gypsum board.
i.e. heavy, stiff walls.

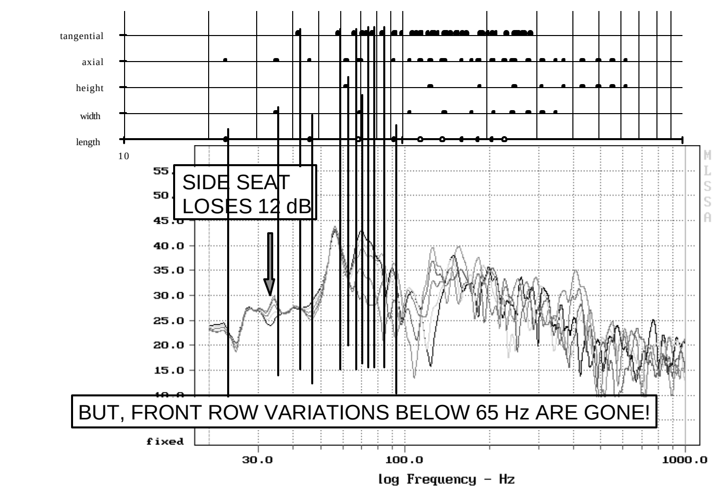
215

As stated earlier, heavy, stiff walls absorb little energy at low frequencies. This helps to keep the bass energy inside the room, and the poorly-damped standing waves will exhibit strong high-Q peaks and dips, and ring energetically. Also, the reduced absorption at each reflection means that tangential modes may join the axial modes as being factors to deal with.



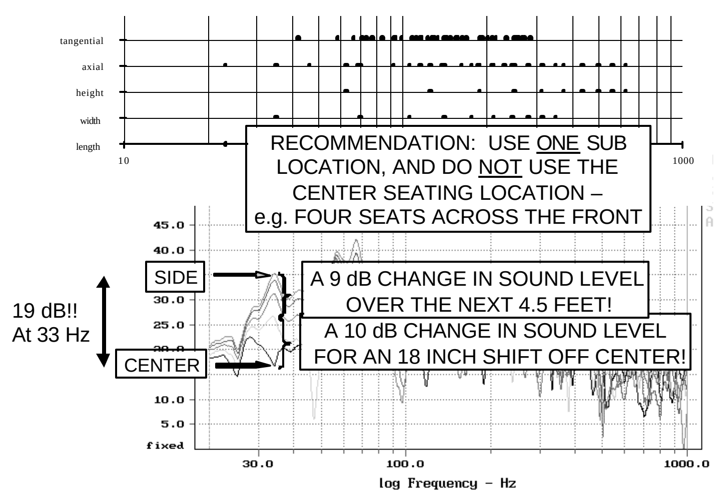
variations in front row – **one front sub**. 217

In this room, it is clear that the tangential modes are very active. Note that it is possible to identify almost all of the axial and tangential modes in the visible peaks at low frequencies. Moving across the front row produces a rather large 19 dB variation from center to side seats over a very important part of the bass frequency range. It would be nice to reduce this.



variations in front row – **two front subs**. 218

Employing the tactic used in the previous example, two subs, one in each front corner, we certainly get consistent performance across the front row, but we have sacrificed a lot of good bass in the process.



variations in front row – **one front sub**. 219

Looking closer, we notice that 10 of the 19 dB change occurs in the first 18 inches away from the center location. This is a high-Q null. The remaining 9 dB accumulate over the next 4.5 feet. Taking advantage of this knowledge, we decide to go back to one subwoofer location – we may in fact use two subs stacked at that location, to get enough power into the room. We avoid the center seating location, choosing to arrange the chairs symmetrically around the room center line.

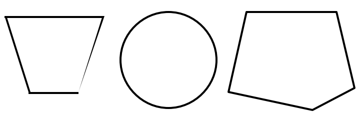
None of this would be obvious without measurements – good measurements, and enough of them to help design the best listening experience for the circumstances.

Some Observations

- **You cannot escape from room modes!**
- **ONLY high resolution measurements can help you understand what is happening. 1/3 octave not enough.**
- **Mode control with multiple subs is <u>very</u> position dependent, but it can work IN CERTAIN ROOMS.**
- **Massive, rigid walls make the situation much more complicated by activating tangential modes.**
- **Spatial averaging of measurements is essential to address the needs of an audience.**
- **You will <u>never</u> satisfy everybody in the audience.**

220

But, You Have Heard That Non-Parallel Walls Solve all of These Problems.



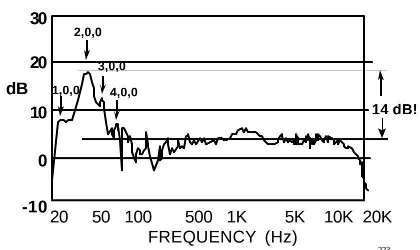
221

What About Non-Parallel Walls?

- The modes are not eliminated
- The strength of the modes is much the same as in a rectangular room
- Because of the complexity of the standing waves, predictions of pressure distributions are not easy, and may not be practical.
- Why bother?

222

A Room With Only One Pair of Parallel Surfaces



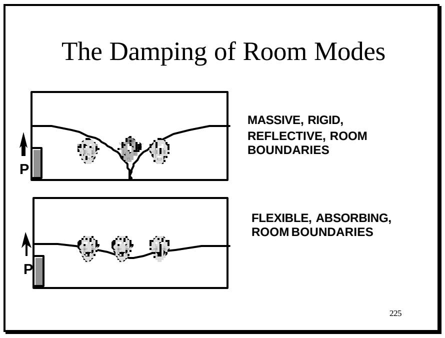
223

Here is an example of a room that had only one pair of parallel walls. The high ceiling was sharply angled, about 40 degrees. The side walls had several large openings, recesses, and a large stairwell. There was hardly any parallel surface left that was at a fixed spacing from one across the room. Taking a simple look at the room, it should have some advantages. But, the bass in that room was awful. It boomed horrendously. I know, because it was in my last house. When measured, it revealed a set of very clear resonances, all of them associated with the remaining pair of parallel walls. With no competition from any other modes, these ones sang their little hearts out.

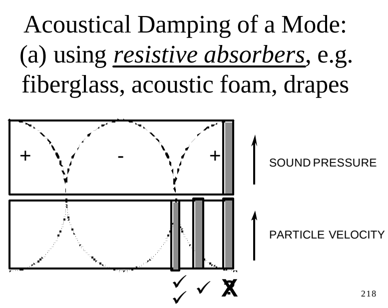
Sound Absorption
reduces the
energy in the modes.
This is called DAMPING.

224

By removing energy from the modes using damping at the reflecting surfaces, it is possible to attenuate the high-pressure regions of standing waves, and to elevate the low-pressure regions. Doing this makes the sound field less variable over the room. In my opinion, this is where we should <u>start</u>, preferably when the room is being built.



Less variation means more happy listeners.



Sound absorption can be achieved with resistive absorbing materials, like fiberglass, acoustic foam, heavy drapes, etc. However, to be effective these materials must be in the regions of high particle velocity. This does not happen immediately at the room boundary, because, by definition, there can be no particle movement when the molecules are hard against a rigid surface. Maximum particle motion occurs in regions of minimum pressure – at the nulls in the standing wave pattern, one-quarter wavelength from the reflecting surface. So, one either needs thicker material, or some air space behind the material. Thicker material works better, but is costly. Either way we face using a lot of space.

Resistive absorbers are most effective when positioned in the high ***velocity*** regions of the standing wave

227

Dropped ceilings, the T-bar systems with lay-in panels actually work well at low frequencies because of the large space above them. The panels themselves, if placed on a hard surface, are no better than the equivalent thickness of rigid fiberglass board purchased from a building supplier. Be careful not to let such ceiling systems buzz or rattle, though.

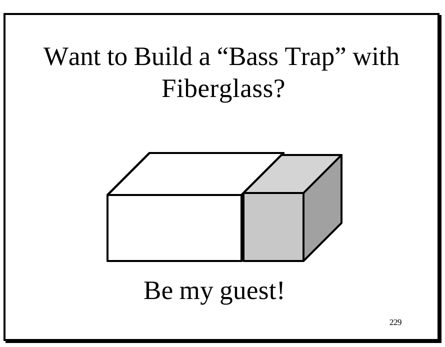
Resistive Absorbers are not Practical at Low Frequencies

- 1/4 wavelength at 100 Hz = 2.8 ft.
- 1/4 wavelength at 50 Hz = 5.7 ft.
- 1/4 wavelength at 30 Hz = 9.4 ft.

228

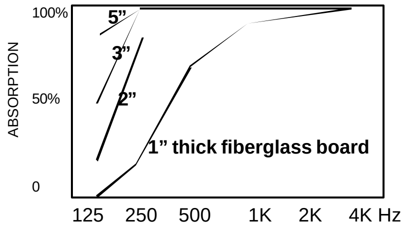
"Bass Traps" are a favorite topic in pro audio. The name conjures an image of a device that seeks out and sucks up bass, never to spit it out. Well, reality is much less romantic, and real bass absorbers are only as effective as space and the budget permit.

If we try to damp really low frequency resonances with resistive absorbing materials, success is possible only if serious amounts of real estate are devoted to the task!



This should be a real stimulus to the building industry, if every home theater had to be about 20% larger in every dimension. Remember, to absorb energy in the modes in all directions, one of these appendages must be built on the ceiling, and another on a side wall. Cost conscious customers might wonder if there is a better way. There is.

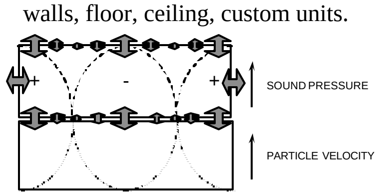
However, Resistive Absorbers Work Superbly at Mid & High Frequencies.



230

However, at higher frequencies, where wavelengths are shorter, such materials work just fine, and are highly recommended. Just don’t cover them with a non-porous material, because such materials need to “breathe”. Resistive absorbers do their work by making it difficult for the air molecules to move around within the “fibrous tangle”. A good low-cost covering, available in a wide range of colors, is polyester double-knit, often used for speaker grilles. If the covering needs to be fire rated, look elsewhere, and pay much more.

Acoustical Damping of a Mode:
(a) using *membrane absorbers*, e.g. walls, floor, ceiling, custom units.



231

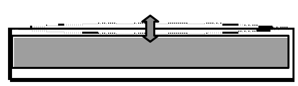
At very low frequencies, we need another technique. It is membrane, or diaphragmatic, absorption. In this, the fluctuating sound pressure of sound causes a surface to move, and in doing so, transfers energy to the moving surface. So, when you feel the bass in your feet, or feel vibrations in the walls, you are experiencing membrane absorption. Obviously, the best location for a membrane absorber is in a high-pressure region for the problem mode. If the room is constructed with bass absorption in mind, the problem is diminished from the outset. “Let the boundaries move” could become a mantra for enlightened room design.

Membrane (mechanically resonant) absorbers are most effective when located in the high ***pressure*** regions of the standing wave pattern.

The room boundaries do it naturally.

224

Mechanically Resonant Membrane Absorbers

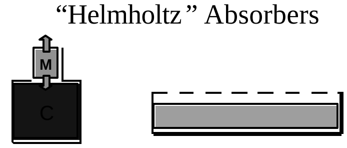


The resonant frequency is determined by the moving mass of the exposed panel, and the compliance of the air inside the enclosure (the volume/depth). A heavier panel, or a deeper box, reduces the frequency. Damping is achieved by mechanical losses in the panel material, and by acoustical losses in the fiberglass. Lossy, “soft” panels have low-Q, i.e. absorption over a wide frequency range. Heavy vinyl has been used successfully.

225

Commercial membrane absorbers are available. Most are flat, and fit against the room boundaries. Others are designed to fit in corners, or to stand freely. Be sure to check that they exhibit high absorption coefficients at the frequencies you wish to damp. Some of them work well at lower mid frequencies, but lose it in the deep bass region. Ironically, a single layer of wallboard on standard studs does a decent job. Things get problematic when a second layer of wallboard is added, or bigger studs are used, or the wall is filled with sand, etc. A somewhat compliant inner surface to a room is a generally good idea. “Let the boundaries move.” Mind you, a wall or ceiling that rattles and buzzes is not welcome. So pay attention to how it is constructed.

Acoustically Resonant “Helmholtz” Absorbers

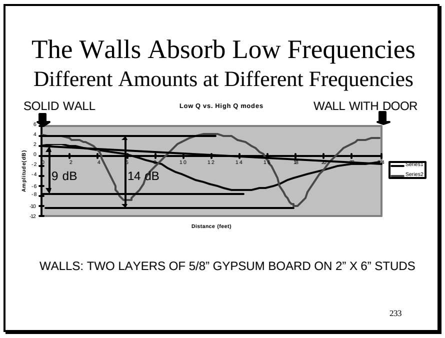


In a “classic” resonator the resonant frequency is determined by the mass of the air in the neck, and the compliance of the air in the chamber (volume). This is the “soda bottle” resonance.

This can be expanded into a surface, with the resonant frequency being determined by the mass of the air in the slots between the bars, and the compliance of the air in the cavity behind. Some fiberglass in the cavity provides damping.

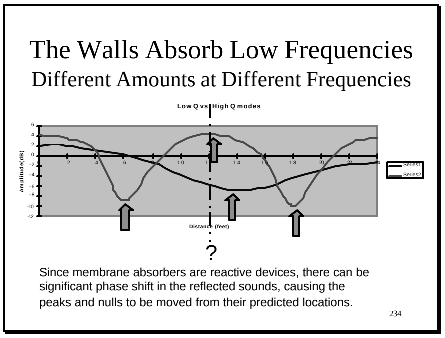
226

A second kind of low-frequency absorber involves a tuned acoustical resonance. Usually these utilize slats with spaces between, and a damped volume behind.

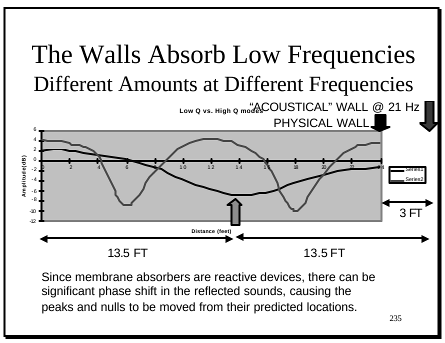


If a room surface absorbs sound, it will do so differently at different frequencies. It is the nature of such surfaces to slightly alter the timing of the reflected sounds (phase shift) depending on the frequency of the sound relative to the preferred frequency for the absorbing surface.

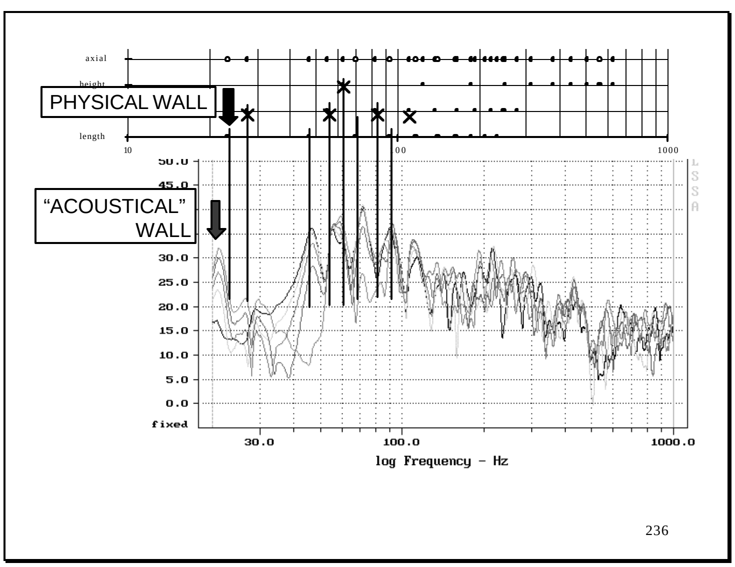
A real world example of this is shown here. It was noted that one wall of a room was vibrating quite actively. The wall had a door in it, and it appeared that this reduced the structural stiffness, allowing much more movement than the other walls. Out of interest, we measured the pressure distribution across the room for the first two modes. The first order mode had more damping than the second.



This is seen as a lower maximum-to-minimum pressure variation – 9 dB for the first-order mode vs. 14 dB for the second. Also apparent was a positional shift of the high and low pressure points. They were not exactly at the locations simple mathematics would predict. The null of the first order mode was substantially off center, towards the wall with the door.



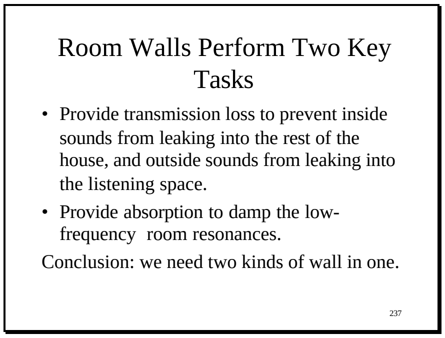
I made the simple assumption that most of the absorption was occurring in the wall with the door, and speculated that the phase shift in the absorbing wall was making it look (acoustically) as though it was farther away. On this basis, I took the distance from the "solid" wall to the null to be the true ¼ wavelength, and projected the same distance towards the wall with the door. This put the "acoustical" wall three feet away from the real one! Acoustically, <u>at this frequency</u>, the room was behaving as if it were three feet longer than the physical length.



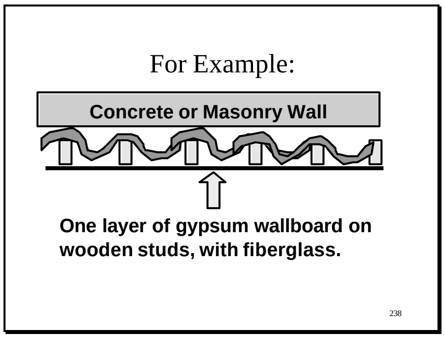
If this is true, then the frequency of the first-order resonance must not be at the frequency predicted by a measurement of the room length, but at a frequency appropriate for a room three feet longer.

This graph shows that this is so. At the frequency calculated for this mode based on the room dimensions, there is no resonant peak. However, at a frequency appropriate to a room three feet longer, there is a healthy resonant peak.

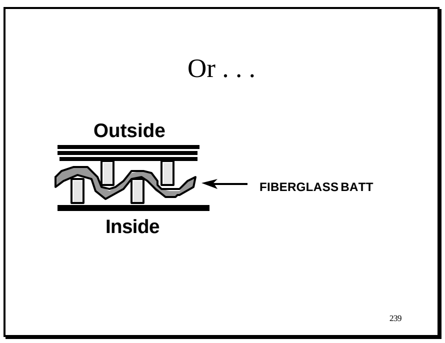
This explains why, very often, it will be found that the measured resonances do not exactly correspond with the calculated ones. This is also why room acoustical modeling programs do not always give the right answers. In rooms it is <u>essential</u> to make high-resolution, accurate, measurements in the room <u>after</u> it is built.



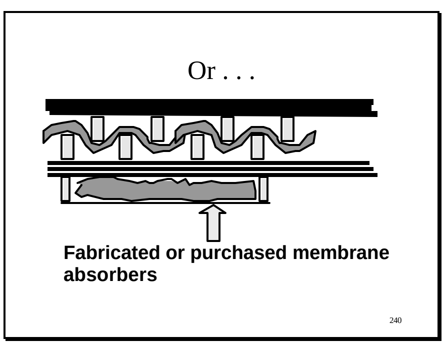
Sound transmission loss in a wall is a measure of how well it prevents sound from traveling through it. Absorption coefficient is a measure of what proportion of the sound falling on a surface is absorbed, and not reflected back into the room. They are two very different things, and they are often confused. Most studies of transmission loss were concerned with privacy, and focused on speech frequencies, not paying much attention to low bass frequencies. With today's powerful woofers and subwoofers, we need new and higher standards.



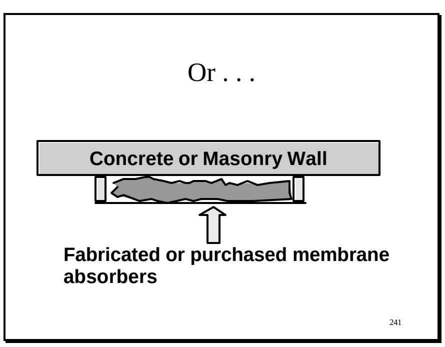
Here the massive, stiff, concrete is providing the bulk of the sound isolation. The standard stud wall provides some attenuation, as well as some sound absorption. Its vibrations are isolated from the concrete wall by an air space and, at low frequencies, the larger the better. Some fiberglass in the cavity adds to the effectiveness by damping reverberation in the space between the walls, not by actually preventing sound from passing through it.



Here is another dual-purpose wall, with multiple layers of gypsum board substituted for the concrete. Again a large part of the isolation is due to the mechanical separation between the inner "microphone" wall and the outer "loudspeaker" wall. If the walls are connected, the "microphone" talks directly to the "loudspeaker" and there is almost no sound attenuation. Therefore, these walls must not be mechanically connected at any point – no conduits, no water pipes, nothing to link the inner and outer walls. HVAC penetrations must be flexible, and be equipped with special acoustical absorbers on each side of the wall to prevent sound leakage through the duct itself.



If acoustical isolation is absolutely paramount, then there is little choice but to go for broke. Build a really massive double wall, and then add membrane absorbers on the inner surface. Always remember that low-frequency isolation can be defeated by even quite small air leaks. Keep an eye on the builder, and be sure that <u>all</u> joints are caulked, and that <u>no</u> penetrations or mechanical short circuits have crept in.



A membrane absorber can be considered a mechanically resonant panel absorber. Another type of low-frequency absorber is the acoustically resonant absorber, also known as the Helmholtz absorber. In these devices, one or a few large openings, or many small openings, resonate with a volume of air at the appropriate frequency. These require careful design, and careful construction, but can work very well.

The MOST PROBLEMATIC rooms I have ever encountered are those that have been constructed with massive, stiff walls.

They have:
1. High Q resonances that ring
2. Poor bass uniformity over the listening area.

242

It is truly sad when someone has spent good money building a room of "solid" walls, based on the belief that it keeps the bass in. Yup, it sure does, but it makes that bass unpleasant to listen to. To those who suggest that thumping a wall with a fist is a useful test of a wall, I will say that it is about as useful as the knuckle test is of a loudspeaker enclosure. Both are highly unreliable because sound waves energize room walls and loudspeaker enclosures very differently than fists or knuckles.

The Rules for Good Sound in Rooms

At middle and high frequencies:
- ✓ Start with a loudspeaker that was designed to function well in a variety of different rooms.
- ✓ Use geometry, reflection, diffusion, and absorption to achieve good imaging and ambiance.

At low frequencies:
- ✓ Maximize the output from the subwoofer(s).
- ✓ Achieve a uniform performance over the listening area.
- ➢ Equalize to achieve good performance.

244

So, we have done what we can to achieve uniform sound quality over the listening area. Now we will attempt to make it sound good by using equalization.

Equalization! You thought that it didn't really work. That equalizers added phase shift and other ugly stuff.

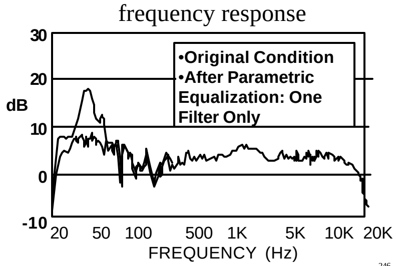
Historically, equalization has not lived up to its promise. However, we have learned what it can and cannot do. We now understand why, the way it has been traditionally done, equalization we doomed to disappoint.

The key to "intelligent" room equalization is in knowing what can and cannot be corrected with filters.

245

Remember room resonances, those things that cause "boom" , "hangover" , "ringing" , "coloration", and that mask our ability to follow bass melodies. They behave as minimum-phase phenomena, and they can be equalized if the measurements have enough resolution to show the details of what is happening, and if they resonances are addressed with carefully matched parametric equalizers.

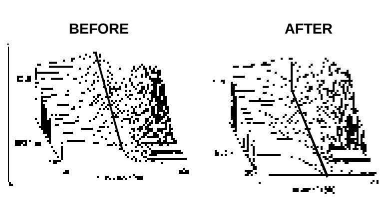
Parametric equalization fixes the frequency response



246

Here is a room with a really big boom. I am familiar with this one, as it was in my last house. It was where I began experimenting with selective parametric equalization, as an alternative to moving the furniture around or engaging in massive reconstruction.

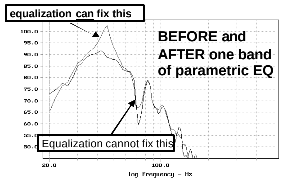
And the transient response is also fixed!



And it really works. Kick drums that previously boomed relentlessly, became "quick" and "tight", and bass guitarists could actually be heard playing harmonies. Organ pedal notes were all there in proper proportions. An amazing transformation.

It doesn't work everywhere in the room, but nothing does. Those pesky standing waves are still there. They will not go away.

A problem and solution:
in the frequency domain



248

This is one I showed earlier, pointing out what can and cannot be equalized.

Dips are caused by acoustical cancellations

- The problem is ***very*** position sensitive – it will be different at different positions in the room. In fact that leads us to the solution: move the speaker, the listener, or both. A few inches may be sufficient.
- Trying to fill the dip is foolish! A 10 dB boost uses 10x more power! A 20 dB boost uses 100x more power! You have just added a 10 or 20 dB resonance that will be clearly audible at most locations other than the one at issue.

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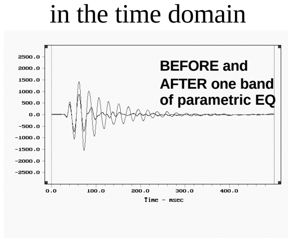
And this is why.

A problem and solution:
in the time domain



250

Just to confirm that our ears weren't lying to us, the time-domain measurement shows that the extended ringing of the original room is tamed.

So, equalization can work!

- If you have high resolution measurements
- If you try to fix the right things
- If you have a parametric filter that you can tune to match the problem.

But, what happens if you don't?

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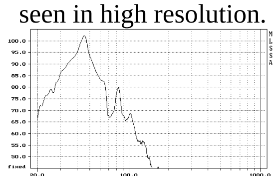
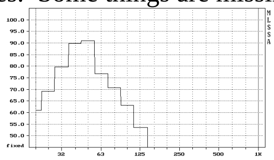
This is the problem situation, seen in high resolution.



252

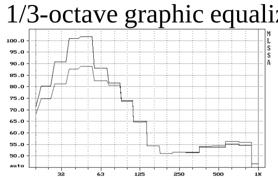
This what a 1/3-octave analyzer sees. Some things are missing!



253

It is amazing that so much information could just slip away so easily.

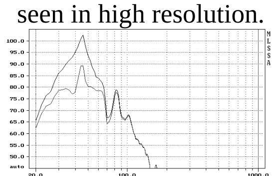
Oh well, let's do something with the 1/3-octave graphic equalizer.



254

However, we do have a "graphic" equalizer, so we had better do something with it. [A "graphic" equalizer is the one with a large collection of fixed-bandwidth filters, adjusted using a collection of sliders on the front panel].

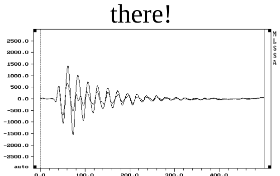
This is what we <u>really</u> did, as seen in high resolution.



255

When it was done, the customer complained that the bass level was lower, and it still was boomy. This is why. When we look at things with the clarity of high-resolution measurements we see what happened. The crude measurement did not show the true situation, and the crude equalizer could not address it even if we could see it.

Less bass, and the ringing is still there!



256

And, yes, the time domain was still corrupted, because the frequency domain was still corrupted.

It is no wonder that equalization has acquired a bad reputation.
It deserves it!

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Bass Equalization can work!

- If it is based on accurate high resolution measurements - i.e. not simplistic 1/3-octave real-time analyzer data.
- If the equalization addresses the room resonance problems with accurately matched parametric filters - not the simplistic 1/3-octave multifilter devices.
- If you do not try to fill dips caused by acoustical cancellations. It is impossible.

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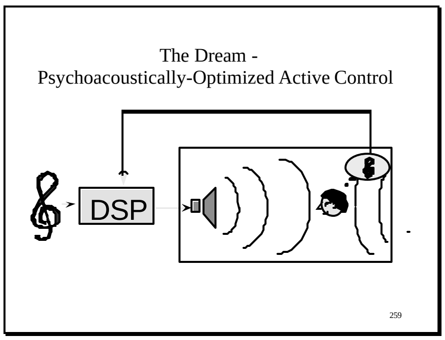
To see what is happening at low frequencies, analyzers must have resolution of 1/10 octave or better.



One day we will be able to have all of this done automatically and 'perceptually correct". The holdup is partly cost, and partly the development of the measurement and equalization methodology that is smart enough to know what is wrong and what it should and should not try to fix. It has to be foolproof.

Sadly, equalization cannot fix everything. Some problems will remain stubbornly in the acoustical domain, so there will be a continuing role for people who understand room acoustics, and who provide the all-important final link to the customers' satisfaction.

Some relevant scientific publications by staff of the Harman International R&D Group (F.E. Toole, S.E. Olive)

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4. **F.E. Toole, "Loudspeaker Measurements and Their Relationship to Listener Preferences", J. Audio Eng, Soc., vol. 34, pt.1 pp.227-235 (1986 April), pt. 2, pp. 323-348 (1986 May).
5. **F.E. Toole and S.E. Olive, "The Modification of Timbre by Resonances: Perception and Measurement", J. Audio Eng, Soc., vol. 36, pp. 122-142 (1988 March).
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9. F.E. Toole, "Listening Tests - Identifying and Controlling the Variables", Proceedings of the 8th International Conference, Audio Eng, Soc. (1990 May).
10. F.E. Toole, "Loudspeakers and Rooms for Stereophonic Sound Reproduction", Proceedings of the 8th International Conference, Audio Eng, Soc. (1990 May).
11. S.E. Olive, "The Preservation of Timbre", Proceedings of the 8th International Conference, Audio Eng, Soc. (1990 May).
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16. ******S.E. Olive, P. Schuck, S. Sally, M. Bonneville, "The Effects of Loudspeaker Placement on Listener Preference Ratings", J. Audio Eng. Soc., Vol. 42, pp. 651-669 (1994 September).
17. F.E. Toole and S.E. Olive, "Hearing is Believing vs. Believing is Hearing: Blind vs. Sighted Listening Tests and Other Interesting Things", 97th Convention, Audio Eng. Soc., Preprint No. 3894 (1994 Nov.).
18. S. E. Olive, "A Method for Training of Listeners and Selecting Program Material for Listening Tests", 97th Convention, Audio Eng. Soc., Preprint No. 3893 (1994 November).
19. S.E. Olive, "Separating Fact From Fiction Through Listening Tests", proceedings of the 1995 DSPx Technical Program, pp. 454-463, (1995 May).
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33. F.E. Toole, "Sound Reproducing Systems", in McGraw-Hill Encyclopedia of Science and Technology, ninth edition, in press.

**** For these papers the authors received Audio Engineering Society Publications Awards**

In 1996 Floyd Toole was awarded the AES Silver Medal "for significant developments in the subjective and objective evaluation of audio devices".

For some lighter reading: go to www.harman.com . Look under "white papers".

Useful References:

"The Master Handbook of Acoustics", Fourth Edition, by F. Alton Everest
McGraw-Hill, 2001

This comprehensive, and readable text goes beyond the normal "handbook" formula, and tries to give the reader some understanding of why things happen. For those interested in low-frequency absorbers, there are several helpful sections.

"Loudspeaker and Headphone Handbook", Third Edition, edited by John Borwick.
Focal Press, 2001

Is there a definitive book on this subject? This may be it. Up to date and loaded with detailed engineering level information.