Designing Home Theaters and Listening Rooms: Part 1—Acoustical Perspectives

by

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OUTLINE

The 500-page book *Sound Reproduction—the Acoustics and Psychoacoustics of Loudspeakers and Rooms*, by Floyd E. Toole, Focal Press, 2017 has in-depth discussion and data on many topics. In this much-simplified guide readers will be provided with some additional information, and directed to parts of the book, using figure and section numbers and page numbers for more explanation.

Other documents on this website will also be referred to. The website will change with time, so updates to this guide will follow; check the edition date at the end.

1 Introduction

Designing a custom home theater is often as much a matter of visual aesthetics and atmosphere as it is sound or video quality. High levels of ornamentation are no assurance of good sound, although expectations will undoubtedly be high. Getting good sound is a separate exercise, and few interior designers are aware of, much less sympathetic to, acoustical needs. One is sometimes presented with a visually attractive—depending on individual taste—design at the outset. The large screen is featured, but all else is expected as if by magic, to be massaged to fit within the visible shell. In some designs, most of the interior surfaces are covered with fabric, which hides much ugliness, but which can contribute to making the theater overly dead unless the fabric is acoustically transparent. The loudspeaker and acoustical requirements need to be conveyed to the interior designer at the earliest possible stage. There may well be “negotiations,” but when the lights are down, the ornaments fade, but the sound remains. Let it be a challenge to the creativity of the designer to deliver a pleasant visual design that does not compromise the sound or the picture. For the interior designer, the theater is the product. For audio people, the sound is the product. Compromises may be necessary.

For many designers and owners, it is assumed that the loudspeakers should be invisible. This is not always possible, and if the highest level of performance is desired, it may not be desirable. Industrial-strength appliances are somehow fashionable in domestic kitchens, where they are probably not necessary, but loudspeakers that provide beautiful music of all kinds and an essential component to movies must be invisible? Obviously, I’m biased!

The video side of theatrical presentations tends to be under excellent control when displays are professionally calibrated to be spectrally “neutral.” This guide focuses on what is necessary to deliver neutrally balanced sound—a transparent “window” into the audio that is recorded. This is what listeners have for decades told researchers that they like. However, what we see and hear are both dictated by the program content, and that is not always of the highest standard. The world’s perfect audio/video system is still at the mercy of recorded programs. When they are good, the results can be enormously impressive.

Although there is a lot of discussion about custom home theaters, the reality is that few people can afford them, and not everybody who can afford them wants them. The good news is that, with few compromises, very high-quality video and audio are possible in multipurpose rooms—where we live. Acoustical needs are real, but they can often be achieved using “stealth” techniques that still leave a visually attractive room when movies are not rolling. There are choices to be made.

Some people think that home theaters are the “poor cousins” of cinemas. For the bulk of the population this is unfortunately true. Small, low budget systems can be highly entertaining, even though they may fall short of the highest goals. Most people cannot generate, or their neighbors will not tolerate, the thundering bass and high sound levels experienced in cinemas. Nevertheless, thanks to progress in loudspeaker design and manufacturing, even modest systems can approach the sound quality of “big ticket” systems. Paying more is not an assurance of higher sound quality, although industrial design and sound output may be more impressive.

Those of us who have experienced state-of-the-art home theaters know what is possible. Stereo music, video music concerts and movies can be spine-chillingly good. However, the irony is that the sound in many cinemas and some dubbing stages is not impressive; not something to be emulated in our homes. Recent surveys have indicated that, due to costs, a high percentage of movie sound is mixed in home-theater-sized facilities, using cone and dome loudspeakers. This could be a good tendency from the home entertainment perspective; however sound quality in movies is not reliable, as a result of unfortunately ill-conceived and poorly executed calibration procedures. Chapter 11 explains the situation. Years ago, I would sometimes play a good music CD after watching a movie, just to confirm that I had just spent the better part of two hours listening to mediocre sound.

The book discusses loudspeakers in detail, concluding that if they are chosen with care the rest of the task becomes much simpler. Room EQ cannot turn a sow's ear into a silk purse, to recycle an old expression. In fact, if properly designed loudspeakers are chosen, room EQ should be avoided at frequencies above about 500 Hz because there is a risk of degrading them—the book explains this in detail (Sections 13.2.3 p. 371, 12.2.3 p. 348, 4.6.3 p. 84). Two ears and a brain are much more analytical than a small microphone and a spectrum analyzer. Much of what we perceive involves a cognitive component that is totally missing from steady-state in-room measurements.

In movie sound the center channel is the workhorse—the most important loudspeaker in the room—yet in some systems it is almost an afterthought, occasionally even omitted. Similarly, the surround loudspeakers, which do much less work, are no less important when it comes to the requirement for sound quality excellence. Budget and space limits force many compromises, but if it is possible, the surround and elevation/immersive loudspeakers should be designed to the same sound quality standards as the front L, C, and R loudspeakers. The ideal situation is that the surround and immersive loudspeakers are smaller versions of the L, C and R loudspeakers, mounted so that the prime listening location receives direct sound radiated on axis or at least within the listening window of the loudspeaker (i.e., not more than 30 degrees off axis. The latter is something that in- or on-wall designs compromise because the direct sound arriving at the prime listener is sometimes a far off-axis response. Done properly, the result is a seamless sense of space and envelopment, but obviously this presents challenges in a space where loudspeakers are intended to be "invisible." If spinorama or similarly comprehensive data are available, brands and models can be mixed if they exhibit close similarity. This opens up alternatives, but nevertheless, an adjustment of aesthetic priorities may be required.

We are much more critical of sound quality when listening to a single channel, which is why subjective loudspeaker evaluations should be conducted in mono (see Chapter 3 and Section 7.4.2, p. 174). In movies and television the center channel is often operating alone—monophonically—as do other channels from time to time, and at those times both the loudspeaker and the room around it are under close scrutiny.

Stereo reproduction is built around mono left, mono right and double-mono for amplitude-panned phantom images, including the featured artist in the center location. Only if there are uncorrelated captured or synthesized room ambience sounds is the situation more complicated. The result is that the L and R loudspeakers matter a great deal, and the room contributes to the effect . . . but, its contribution is most often one of redirecting off-axis radiations from loudspeakers in the direction of listeners. If those sounds leaving the loudspeaker were not timbrally desirable, we may be better off not hearing them, and the best option is to absorb or attenuate them. So, why not start with well-designed loudspeakers?

In terms of the "soundstage and imaging" consequences of early reflections there is much to be said (Chapter 7). There is significant variation in the results of investigations, whether one relies on consumer audio or professional audio experimental data and anecdotes. Some may be surprised to learn that the dominant factor in stereo "soundstage and imaging" is the recording itself (Section 7.4.2, p. 174). It is also clear that personal preference and habituation to certain circumstances are also factors, which explains personal preferences for some recordings over others, and notions about acoustical treatments that provide the greatest rewards. Some of the advice about acoustical treatment comes from sources with a financial interest, for whom "more" seems always to be "better," and "what" is "what they sell." It is not always wrong, but a questioning attitude is advised.

At this stage in the evolution of audio, listeners should be encouraged to embrace multichannel audio for music as well as movies. There is *no* reason why the L and R loudspeakers in a multichannel system cannot deliver state-of-the-art stereo. Having additional channels and loudspeakers in the room allows for the possibility of tasteful upmixing of stereo programs to enhance the sense of envelopment that only multiple channels can deliver persuasively. Some of the widely available upmixers are, in my opinion, overly enthusiastic about sending sound to the center and surround channels and in disrupting the front soundstage. However, there are some that please me, and today's receivers and surround processors are flexible, placing the key parameters under the listener's control. Experimentation is possible, and nothing is permanent.

Loudspeakers and rooms function as systems—both are involved in what we measure and hear in rooms—but nowhere is it more obvious than at bass frequencies, where room resonances dominate what is heard. Chapters 8 and 9 address the issues of delivering bass to listeners in small rooms. Bass accounts for about 30% of the perception of overall sound quality, so getting it right is important.

2 Some Practical Considerations

This guide focuses on loudspeakers, room acoustics, and how they affect sound quality. However, there are other considerations, some of a purely practical nature, that limit what one does. Most listening spaces are created within already existing homes; it is a rare luxury to be able to start with a clean sheet of paper and a large budget. When this is possible, the first concern is the size of the audience, which determines the size of the seating area and from that the essence of room dimensions and proportions. Seats around the perimeter should be well removed from room boundaries—1 to 2 m (3 to 6 ft) if possible. Listeners in seats near the side walls may experience excessive bass and will experience reduced "envelopment." They may also be distracted by localizing the side surround loudspeakers (Section 15.7.1). Space around the listening area is greatly advantageous.

The size of the screen can affect loudspeaker placement if it is not acoustically "transparent." Placing loudspeakers to the sides and bottom (or top) of the screen is frowned upon by cinema purists, but the arrangement can work because the "ventriloquism" effect is very strong—we tend to localize to the visible moving lips, door slams, etc. A short time with such a system and adaptation sets in. The center loudspeaker delivers much of the on-screen sound, and localization errors are normally not noticed. This happens even in the horizontal plane where we are especially sensitive to such things, including in cinemas.

If foreground stereo listening is desired, there is no reason why the front L and R loudspeakers cannot be "audiophile" quality, and placed outside the screen boundaries to avoid any concerns about screen losses. There are now some products than can satisfy fastidious music listeners and play loud enough to thrill movie watchers, although such loudspeakers can be large and expensive. Properly implemented bass management with single or multiple subwoofers means that the L and R loudspeakers, even full-range floorstanding models, can be designated “small” in the setup routine, meaning that they can probably play louder than in their full-range mode. When maximum sound levels for loudspeakers are specified, it is normally assumed that the signal is broadband, and the limit is often imposed by the displacement of the woofer at very low frequencies. High-pass filtering, as happens in bass management, results in reduced distortion and higher output capability from the woofer. The limitation in output then moves to the tweeter, or other drivers. Low-slope crossovers have an unwarranted appeal to some customers because the promoted characteristic, linear phase, is not audible (pp. 91–93). But they are a liability in terms of maximum output because drivers are forced to absorb energy at frequencies outside their normal operating bandwidths.

Figure 1 shows my solution for the way we live our lives, a multi-purpose room with a minimum of compromises in things that matter. Large floorstanding loudspeakers can dominate a room, so here the large powerful audiophile L and R loudspeakers are elevated and inverted, thereby greatly reducing the visual impact. The deliberately irregular front wall (0.76 m, 30 in depth variations) provides visual interest, performs as a sound scattering surface (Section 4.10.4) at lower frequencies and lessens the adjacent boundary issue for the wall-mounted loudspeakers (Chapter 9). Having both direct view and projection display options is very convenient in this room where we read, converse, listen to music, watch TV and, at the push of a few buttons, find ourselves in a darkened, acoustically damped, home theater—at any time of day. The system is capable of more than adequate sound levels, and provides impressively neutral reproduction in stereo or multichannel modes. Overhead immersive loudspeakers have yet to be added. For practical reasons (foot traffic) a hard floor surface is adjacent to the front wall. The floor reflection point for the center channel tweeter is within the carpeted area, but it is interesting to note that floor reflections are less audible than measurements indicate (Section 7.4.7, p.193). A Sound Field Managed four-subwoofer system removed any need for bass traps, which would have significantly altered the visual appeal of the room.

Figure 1. The author's entertainment/family room showing the dashed outline of the 10 ft motorized front-projection screen. The Revel Salon2s are inverted, tweeters at the bottom placing the soundstage at the right elevation. A Revel Voice2 is under the 65-inch flat-screen monitor with the tweeter as high as possible. The surround loudspeakers are Revel Gem2s, themselves excellent stand-alone loudspeakers. Four JBL HTPS-400 subwoofers in a Sound Field Managed arrangement provide bass (see Section 8.2.8, p. 244). See also Figure 7.21, p. 192.

For dedicated movie viewing, the cinema practice of placing loudspeakers behind a perforated or woven screen is the tradition in dedicated home theaters as well. Correction for any high frequency attenuation by the screen is easy and tests have shown that there is minimal, some say no, audible degradation (pp. 300–301). Woven screens have little acoustical loss, but some at least have visual tradeoffs. Placing the L and R loudspeakers outside the screen area is a safe alternative, which, in the author's experience, involves no compromise in the entertainment value of movies. This is a personal opinion, not a scientifically verified observation, and others may disagree. This option is, of course, available only when the screen and room widths allow for it.

The best possible sound should be experienced at the "money" seat, where the delays and sound levels are calibrated during setup. If the customer likes listening to stereo music, this seat should be at the apex of an equilateral triangle. (The distance between the L and R loudspeakers ideally should be the same as the distance from each of them to the listener's head.) This puts the L and R loudspeakers at the recommended ± 30° angles from center. There are rooms in which this is not possible, so it is fortunate that this angular separation can be reduced slightly with little sacrifice. It is not as important for movies, so the prime listening location can be moved forward or back, as required, to provide the customer with the best experience for the chosen entertainment, stereo music or multichannel movies. The prime listening location(s), and as many other seats as possible, should be on the centerline of the room: equidistant from the side surround loudspeakers. This requires a progressive elevation of seating rows moving towards the rear of the theater.

Some people are concerned about the centerline of the room having a problem null (the first order width mode). It can happen if there is only one source of bass, but even stereo has two (see Figure 8.13, p. 236). The problem is automatically eliminated in multi-sub installations.

Many of the traditional rules dictating optimal viewing distances for different screen sizes are changing as picture quality improves. As pixels become vanishingly small, the decision now includes the perspective one is most comfortable with. Personal preference is a factor.

3 Sound Isolation and Background Noise Reduction

Noisy movies and music can be disruptive to activities elsewhere in the house, and noisy activities elsewhere in the house can be disruptive to quiet passages in movies and music. It is important to discuss this with the customer, because sound isolation can be expensive, and in retrofit situations it may not be possible to achieve ideal solutions.

The PowerPoint tutorial “Sound Isolation and Noise Control in Home Theaters” on this website provides basic guidance about objectives, methods and materials. This is one area where a competent acoustical engineer may be necessary if effective sound isolation and a quiet background are important. Acoustically treating the interior of a room is simple and inexpensive by comparison. In new construction, if there is any opportunity to move noise sources away from the theater room, this can provide the most effective solution of all.

4 The Room at Low Frequencies

As Dr. Sean Olive’s research showed, bass quality accounts for about 30% of the factors involved in making an overall subjective judgment of sound quality (pp. 135–142). It is worth some effort to get it right.

Consider that the bass heard by listeners is determined by:

1. The loudspeaker frequency response at low frequencies. Subjective preferences favor smooth responses to very low frequencies.
2. The dimensions of the room. These determine the frequencies of room resonances and the spatial distribution of the associated standing waves.
3. The locations of the loudspeakers and/or subwoofer(s). This determines how effectively they deliver low frequency energy to the room resonances.

4. The locations of listeners within the room. This determines how influential individual room resonances are in what is heard.
5. The acoustical performance of the room. Wall construction can significantly damp room modes, and low frequency absorbers are an option.

Let us take these items in turn.

1 Loudspeakers vary enormously, so for simplicity we will assume for the purposes of this discussion that the loudspeaker is a flawless source of sound power down to the lowest frequency of interest. For movies especially, this can extend below 20 Hz. All of the following recommendations apply equally to all woofers or subwoofers. The limitation of woofers in floor-standing full range loudspeakers is that they cannot always be optimally located, and they rarely perform well at low enough frequencies to qualify as subwoofers. Bass management and subwoofers can be greatly advantageous.

2 A widespread belief over the years has been that there are "ideal" room dimensions; proportions of length to width to height that result in better sound. The goal is to achieve a uniform distribution of room resonance frequencies. While there is merit to the idea in very specific applications, the arguments are simply not relevant to sound reproduction (Section 8.1.1, p. 220). This is in spite of the fact that some widely used industry "recommendations" contain this kind of guidance—they are wrong. And, non-rectangular rooms don't eliminate the problem, they just make it more difficult to analyze (Section 8.1.2, p. 222). When one employs multiple subwoofer strategies in *rectangular* rooms (Section 8.2.6, p. 238) there may be opportunities to deliver more uniform sound to people in an audience by paying attention to room dimensions in the horizontal plane, but for reasons quite different from those that prompted the original investigations into optimum room proportions.

3 and 4 The locations of the loudspeakers and of the listeners are strongly linked when it comes to optimizing a listening situation. Loudspeaker location determines the amount of energy supplied to individual modes and listener location determines how much of that energy is heard (Section 4.10.2, p. 100; and 8.2, p. 224). Understanding these basics is essential in analyzing a given situation. They are also essential to addressing the problem, whether it is by using conventional bass traps or multiple-subwoofer methods.

5 Bass traps remove energy and thereby damp the room modes. They are maximally effective when located in the high-pressure regions of the offending mode(s). Therefore, the analytical method of applying this solution begins with measurements to determine which room modes are causing problems at the seating location(s). Often it is only one or two modes that generate the offending booms.

Figure 4.21, p. 101 illustrates the basics of room modes, and Figure 6.2, p. 150 shows how calculated modes can be associated with measured ones. Use the "room mode calculator" downloadable from www.harman.com (click on "innovation"), and follow the guidance of Section 8.1, p. 216 to determine where bass traps need to be located in order to address the problem modes.

Because all room modes have high pressure points in corners, these are popular locations for bass traps. However, these are not selective, and remove energy in frequency regions where there are no problems for listeners. A targeted approach may be more appropriate.

*There is a fundamental difference between treating the interior of a studio, where musicians are performing, and treating the interior of a room for sound reproduction. In a studio, **all** modes are potential problems because any number of instruments, voices and microphones can be located anywhere within the space. In sound reproduction, only the modes involved in communicating sound from the (fixed location) loudspeakers to the (fixed location) listener(s) are at issue. Furthermore, the modes need only to be controlled during the playback of programs, not at any other time, which allows us the option of employing multi-subwoofer solutions.*

However, success is not possible without the capability of making acoustical measurements. They need not be costly or elaborate; an inexpensive *measurement* microphone, a laptop computer and some free or inexpensive software are sufficient. Even smartphone apps can be useful. Just aim for frequency resolution of not less than 1/6 octave, with higher resolutions being very helpful in difficult situations because individual modes can be clearly seen (see Figures 8.8 and 8.9, pp. 227 and 229). Some programs have default settings that smooth the graph so much that it all looks as though an artist in the marketing department had generated it. As will be seen in Chapter 8, getting good bass may involve some work and some thought, but the end result is very much worth the effort.

One of the early lessons in performing these bass optimization exercises is that the largest, most expensive subwoofer in the world cannot sound perfect in a room because what is heard is filtered through the room resonances. These resonances must be tamed, either by installing bass traps, or by employing multi-subwoofer solutions.

The primary purpose of multi-sub solutions is to reduce seat-to-seat variations, but some can result in significant efficiency gains—higher sound levels from less power. There are two fundamental techniques, those that work only in rectangular rooms (Section 8.2.6, p. 238 and Section 8.2.7, p. 244) and those that work in any room (Section 8.2.8, p. 244). Room resonances are significantly attenuated. Evidence of resonances that remain can be dealt with by equalization, and because seat-to-seat variations are minimized, the EQ benefits several listeners.

However, if there is only a single subwoofer, equalization maximally benefits the listener seated where the microphone is located. All other listeners take their chances.

5 Reverberation and Diffusion

All rooms have some amount of reverberation: the sound that is reflected many times between and among the boundaries. It continues after the sound source stops radiating sound, getting progressively weaker as energy is absorbed. The time taken for the sound to decay by 60 dB (i.e., to inaudibility) is called the *reverberation time* and it is designated RT_{60}, or simply RT. In highly reflective spaces reverberations times can be very long, making communication of any kind difficult. Gothic cathedrals were so reverberant (RT = 5 seconds or more) that religious rituals had to be memorized because verbal communication from the pulpit was unreliable. Music in such spaces was slowly paced to accommodate the time-domain smearing. Concert halls have RTs in the range 1.5 to 2.5 s favoring different styles of music. Conductors may alter the pace and presentation of a work to better suit specific halls. Opera houses require speech intelligibility so they drop to 1 to 1.5 s. Movies rely on highly intelligible dialog and consequently cinemas exhibit RTs of 0.6 s or less in the middle frequencies. It is common to set limits on the frequency dependent variation of RT. It is undoubtedly important in large performance spaces, because reverberant sound energy is dominant over most of the audience. However, in small listening rooms I can think of no reason why it is a serious concern because there is so very little genuine reverberation. The frequency dependent behavior of a few dominant reflection areas (e.g. side walls) does matter, but RT is the wrong measurement with which to evaluate it.

Home theaters and listening rooms typically fall within the RT range 0.3 to 0.5 s. In such rooms speech intelligibility is excellent, and so is the presentation of musical detail (pp. 102 and 201). This is the range of RTs found in normally furnished domestic rooms—carpet, drapes, upholstered furniture, tables, lamps, bookcases, etc. In many cases no special treatment is required to meet this target—Figure 10.1, p. 282. A persuasive non-technical test is to have one person stand near the center channel location and have a conversation with another person wandering around the room. If there is no problem, the objective has been met—forward-firing loudspeakers are usually more directional than a real human talker, and movie dialog is usually at a higher sound level, so the sound reproduction version is likely to be even better.

Nevertheless, dialog intelligibility problems occur in movies or TV. "Talking heads" on TV are almost always highly intelligible because the lips can be seen (we all lip read to some extent) and there are usually no distracting sounds unless a panel of opinionated pundits gets into a shouting match. In movies anything is possible. Dramatic whispers and mumbles with no lips visible are common, but the main problem comes from background music and sound effects that mask important dialog. Older and hearing-impaired people have special difficulty, but even young people sometimes resort to subtitles these days. The room itself is rarely an issue, unless there is a discrete delayed reflection from a distant back wall in a very long room.

In highly reverberant live performance spaces, auditoriums and concert halls, the sound field can be described as being significantly *diffused,* sounds arrive at a listening position from all possible directions—it is a chaotic sound field. As reverberation time falls, so does the diffusion in the sound field. In cinemas, small listening rooms and home theaters there is no possibility of a diffuse sound field—there is too much absorption—which is good because it is not a desirable acoustical property of such rooms.

Discussions that include references to "diffuse" loudspeakers (typically referring to bidirectional in- or out-of-phase surround loudspeakers) are misguided. They are simply loudspeakers with wide dispersion. Bipole (in-phase) versions are useful as side loudspeakers where it is necessary to deliver strong direct sound to members of a multi-row distributed audience (Section 15.8, p. 420). If the impression of a relatively diffuse sound field is desired in a program it is created by the recording engineer using the multichannel audio system. The listening room should be neutral in this respect. With multichannel audio, a small room can be made to sound as large as the recording engineer decides, but a large room cannot be made to sound small.

So, reverberation in small listening rooms is much diminished, and a diffuse sound field does not exist - what is it then that we hear? The answer is that what we hear is dominated by the direct sound followed by a collection of early reflections. Measuring instruments may show a RT number, but it is really measuring the decay of a relatively small number of early reflections. Interior room acoustical treatments therefore mainly affect the first (early) reflections of sound from loudspeakers as they interact with room boundaries on the way to listeners.

6 Early Reflections—the Physical Factors

This is a topic that has been the subject of some heated debates over the years, and strong differences of opinion exist on how important they are, and how best they should be manipulated.

It turns out that strategies developed some years ago may no longer be applicable. Things change: loudspeakers get better, multichannel audio evolves, 3D immersive audio appears on the scene, and so on. But stereo is still the dominant format for music, so today's listening spaces have to multi-task.

I am old enough to remember when there were essentially no loudspeakers that were "good" by today's standards. The best aspect of performance would typically be the on-axis frequency response, and off-axis misbehavior was the norm. In a somewhat reflective room, that colored off-axis sound detracted from the listening experience, so much trial-and-error repositioning was part of the installation ritual. Absorbing the off-axis radiation was an option, and some recording control rooms went to the extreme of placing absorbing material over most of the front of the room. It then became unpleasantly dead, so the rear half of the room might be rendered more reflective, with scattering/diffusing devices to give the sound field more spatial interest. Chapter 18 in the book shows examples of numerous loudspeakers with such off-axis flaws, as well as several with much more uniform behavior.

Perhaps the most significant development since those early years is that well-designed modern loudspeakers radiate on- and off-axis sounds with more similar timbres. A loudspeaker with good on-axis frequency response becomes a better loudspeaker if that excellence is continued into the off-axis sound—the reflected sound (Chapter 5). Two ears and a brain can hear the difference.

Today, fashions in recording control room design range from the "non-room" approach in which virtually all reflections are seriously attenuated, through to rooms in which the boundaries are almost completely covered with scattering/diffusing devices. In between lies a multitude of choices, in which a frequent requirement is that the first lateral reflections from the L and R loudspeakers be absorbed. A significant difficulty with current practice is that the absorbing materials, often 25–50-mm (1–2-inch) fiberglass board or the equivalent in slab or sculptured foam, do not absorb all of the sound that arrives from specific angles, as in the case of a side-wall reflection. There is a strong frequency dependency (pp. 168 and 174), and even a very common fabric covering is not acoustically transparent at high frequencies (pp. 168 and

170). Basically, such materials simply turn the treble down, altering the performance of possibly well-designed loudspeakers. If the goal is to eliminate the reflection, the solution is to use much thicker absorbing materials: more than 76-mm (3-inch) fibrous material.

An alternative to absorbing a side-wall reflection is to scatter it with a diffuser, thereby attenuating the portion that arrives at the listener. The sound scattered in other directions is rapidly absorbed in typical rooms. Again, the frequency dependence of the scattering device or surface is an issue (pp. 169–172). Many of the shallow diffusing devices in the marketplace function only at high frequencies, being useful mainly to alleviate hand-clap flutter echoes. To be effective down to the transition frequency (below which room resonances progressively dominate), conventional engineered surfaces (e.g. Schroeder-type) need to be about 200 mm (8 inches) deep, and geometric shapes about 300 mm (12 inches) deep. Check for measurements of the diffusion coefficient as a function of frequency; it should be ideally effective down to about 300 Hz.

These depths of materials and devices are often difficult to incorporate in wall treatments of small rooms, so compromises are common. However, the objective is clear. Get as close to it as you can. Needless to say, if the loudspeakers are well designed, a viable alternative is simply to use a flat reflective wall, or rely on typical room furnishings to provide scattering.

7 Early Reflections—the Perceptual Factors

In the end, what we hear determines what is good. In this case, the effects of early reflections in small rooms are mainly in the category of directional and spatial effects. These are strongly influenced by the recordings themselves, and the expectations of listeners, so casual listening is unlikely to be definitive.

Chapter 7 covers this topic in great detail, including results of several double-blind evaluations, so little needs to be said here. Examining both the consumer and professional sides of the audio industry it turns out that there is no single perfect solution for either side. The common assertion that side wall reflections are inherently damaging to reproduced sound is simply not supported by published evidence from a number of carefully conducted investigations and much thoughtful listening. It really is a case of one size not fitting all. Some of the evidence points to human adaptation as being a significant factor—we may simply "prefer" what we are familiar with. Musical taste matters too, with spacious classical renderings standing in contrast to some "in your face" pop material.

In the earlier editions of this book I included the following illustration, Figure 2, offering suggestions for the distribution of acoustical materials on various interior surfaces. There is considerable freedom to be distinctive, to develop your own "style."

Acoustical materials and devices are essential in custom venues that begin with bare surfaces, but there, and anywhere, they should be employed on an "as needed" basis. Sometimes less is more. I have been in too many home theaters that are excessively "dead." Such rooms can work for multichannel and immersive audio, but are not flattering to stereo program unless upmixing is employed. They are also not pleasant places in which to carry on conversations. This can happen when walls cluttered with acoustical devices and loudspeakers are covered with stretched fabric. If the fabric is not adequately porous, it becomes an absorber, and there can be a lot of it.

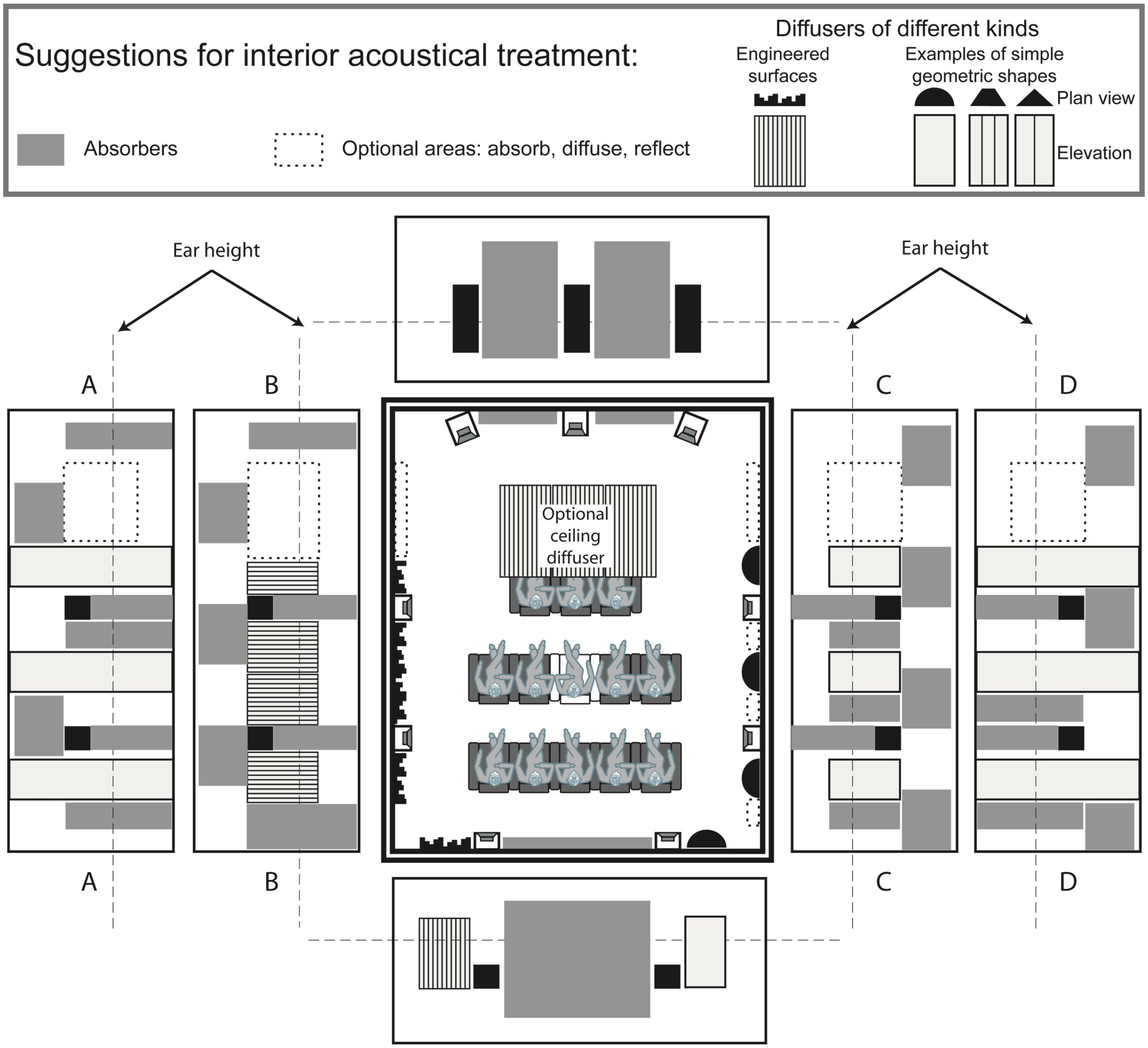


Figure 2. This figure illustrates a few of the many possible ways to combine acoustical materials within a custom home theater. It shows a floor plan with walls folded down to show how materials might be arranged on them. The white seat in the middle is the prime listening location for movies, but the center front row may be preferable for stereo listening. Wall B shows a long array of engineered diffusers in a band situated at ear level. Wall C shows a version that uses semi-cylindrical geometric shapes intermixed with reflecting surfaces and absorbing panels. Walls A and D show mirrored treatments in which the diffusing shapes have been extended floor to ceiling for visual effect. Absorbing panels have been placed in staggered locations to avoid flutter echoes. Many artistic variations are possible and changes will be

necessary to accommodate different numbers of surround channels. The dashed lines that identify the ear height of seated listeners must be adjusted to follow the floor contours of staged seating. This is Figure 22.3 in the earlier editions of the book.

Note that the wall surfaces involved with the controversial first side wall reflections are shown as "optional areas: absorb, diffuse, reflect."

Leaving these areas as flat wall surfaces provides an open and spacious soundstage for those customers who listen in stereo using loudspeakers with well-behaved off-axis performance. If listeners prefer a compact soundstage or have loudspeakers that misbehave off axis, then absorbing the side wall reflections would be appropriate—but do it completely, not partially. With loudspeakers having off-axis problems it will be advantageous not only from a sound quality perspective, but also because reflected sounds that do not resemble the timbre of the direct sound are more likely to be heard as separate spatial events.

Comb filtering is often mentioned in the context of these side wall reflections. It is indeed true that one measures what looks like a comb filter. However, two ears and a brain process sounds in a manner that distinguishes between sounds based on the angle of incidence, a microphone does not. When the direct and reflected sounds arrive from different directions, the perception is normally of a small spatial effect not destructive timbral distortion. Figure 7.3 (p. 164) and the associated discussion are relevant.

An interesting fact is that when we are moving we can hear things that we don't when we are stationary. I have witnessed an acoustical consultant playing pink noise and demonstrating that acoustical interference, which was called "phasiness," was audible when swaying the head from side to side. However, the same phenomenon that was audible in the dynamic situation with pink noise, a highly revealing signal, becomes inaudible if a listener simply walks in, sits down, and listens to music or movies. Such reflections, and there are many of them, fall into the context of "room sound," which human listeners are known to readily adapt to. To a very substantial extent, we are able to "listen through" rooms. It is what happens in live, unamplified music performances, and everyday conversation. In terms of speech intelligibility, most small room early reflections are desirable (pp. 200–201).

When multiple channels are operating simultaneously, these reflections are swamped by the recorded sounds and become neutral factors. In any event, they are not dominating effects, so the choice can be left to the designer/installer/customer. Movable heavy velour drapes on the front side walls can give a stereo listener the choice of absorption or reflection, as the mood or the music demands. Section 7.5, p. 194 discusses the contrast between listening for "business," in a recording control room context, and listening for pleasure at home. It raises some important points.

7.1 Hearing Ability is a Factor

Hearing ability and hearing loss are complex topics. The reality in our industry is that what we individually hear is a product of our own hearing apparatus, and what is in the recordings is the result of what the recording engineers heard through their apparatus. The "apparatus" includes the brain, not just the peripheral auditory organs. Much of what we perceive has a cognitive component—if you believe something, you may just hear it. That is why we do blind tests.

We talk about audiometric tests—evaluations of pure tone thresholds—but these just measure the lowest levels of a few pure tones that we can hear. They do not take into account the fact that above threshold perceived loudness, critical bandwidths, directional and spatial resolution are different for different individuals. We may not hear low level sounds at all, but sounds above threshold might range from relatively normal to uncomfortably, even painfully, loud (hyperacusis). The perceptual dynamic range may be reduced. The increased width of critical bands can change how musical complexities are perceived. We may have difficulty separating sounds in space and extracting information from complex reflective sound fields. Relying on rudimentary audiometric thresholds as a metric of hearing ability is the equivalent to selecting a car by kicking its tires. It is a small part of a complicated story.

Not widely appreciated is that the criteria applied by audiologists relate to speech intelligibility, not hearing musical nuances. The occupational hearing conservation criteria we hear so much about were created for factory and industrial workers. They do not prevent hearing loss. Instead they allow it to happen, attempting only to preserve enough hearing ability that at the end of a working life the factory worker can carry on an imperfect conversation at a distance of one meter. HiFi hearing is long gone. A mid-frequency threshold elevation of 25 dB is considered “normal.” For whom?

When it comes to judging sound quality, listeners within the range of “normal” hearing exhibit significant differences in their abilities to judge sound quality. As thresholds rise, they exhibit increased standard deviations in repeated judgments of the same sounds, and they may also exhibit bias. Why? Because they are unable to hear either the good small nuances, or the bad distortions and colorations. This is something that cannot be improved. There are results from double-blind tests that prove it: Toole, “Subjective Measurements of Loudspeaker Sound Quality and Listener Preferences,” *JAES*, 33, pp. 2–31, 1985; see also Sections 3.2, pp. 36–40 and 7.5.1, p. 196, and all of Chapter 17 in my book. The interesting and at the same time alarming aspect of these findings is that the evidence first came to light while using professional recording engineers and producers as subjects in double-blind listening tests. Hearing loss is an occupational hazard in the audio business.

The most recent (bad) news is something called “hidden hearing loss.” It is a reduction in the ability to separate sounds in space, and to discriminate against unwanted sounds as a function of direction (the “cocktail party,” noisy restaurant effect). This undoubtedly contributes to differences in opinion about stereo soundstages and optimal listening environments, particularly as it involves early reflections. Hidden hearing loss occurs in people, even young people, who do not exhibit elevated thresholds. This is not good.

The weakest link in audio may be “us.” But we can still enjoy what we are able to hear, the way we hear it. I do.

8 Acoustical Surface Treatments

The sounds arriving at a listener’s ears are determined by the loudspeaker and the room:

- the direct sound that is well represented by the on-axis/listening window spinorama curves (Section 5.3, p. 111), followed by
- reflected sounds the timbre of which is dictated by the off-axis frequency response of the loudspeaker, as well as the frequency dependent behavior of the relevant room boundary.

Figure 3 shows some popular options for treating the reflection point of a room boundary: (a) reflection, (b) absorption, (c) scattering by engineered surfaces (e.g. Schroeder-type diffusers) and (d) scattering by geometric shapes (e.g. hemi-cylinders). In (c) and (d) only the scattered components that reach the listener are shown; remaining sounds are radiated in many other directions, depending on the design of the scattering device. (e) shows the distinctive zones in a listening room that may require different kinds and amounts of acoustical treatment, or none, depending on the directivity of the loudspeakers radiating the sound, the dimensions of the room, and the expectations of listeners receiving it. The ray pattern shown is only for the first reflections; higher-order reflections will also contribute to the listening experience. (This is Figure 7.5, p. 167 in the book).

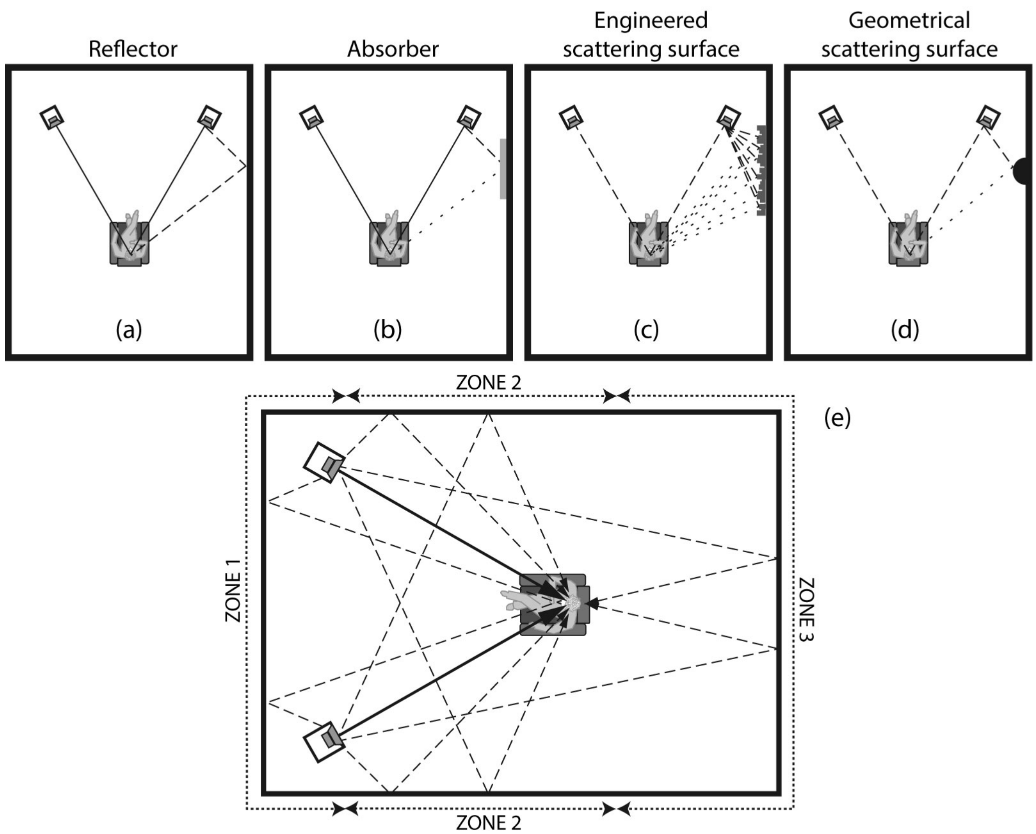


Figure 3.

Of these options, it is important to note that the spectrum of the reflected sound will be altered by the frequency dependent absorption of (b), and the frequency dependent scattering of (c). Reflectors (a) will have negligible effect on the spectrum of the reflection. The same is true for geometrical scattering surfaces (d) except that below a certain frequency they cease to scatter. It is clear that (c) also spreads the scattered sound in the time domain because engineered surfaces radiate sound from all parts of the surface. Perceptually, the time smearing appears not to be an audible effect in normal listening. However, it looks better than a single spike in an ETC (energy-time curve) display (p. 170).

Stereo listeners need to be aware that phantom images, including the featured artist in the center, include a blend of all of the reflected sounds shown in (e). A dedicated center channel loudspeaker is much less complicated.

Below the transition frequency of about 300–400 Hz, room resonances become progressively more dominant, so the issue regarding surface treatments is: how do they perform above 300–400 Hz? Ideally, if one has selected well-designed loudspeakers, the requirement is that whatever absorbing, diffusing or reflecting properties they have, acoustical materials and devices should function uniformly at all frequencies. This is a challenge because achieving absorption or diffusion at lower frequencies requires thicker materials and devices.

The conventional figure of merit for acoustical absorbers is the *random-incidence absorption coefficient*. As the name implies, it is measured in a highly diffuse sound field, with incident sounds arriving from all possible directions. Measurements are made in reverberation chambers. This specification has been useful for predicting the acoustical events in large, somewhat reverberant live music performance venues. But that is not where we listen at home; quite the contrary, our listening rooms are distinctly *not* reverberant and *not* diffuse. However, as things stand, these are the data the industry gives us to work with.

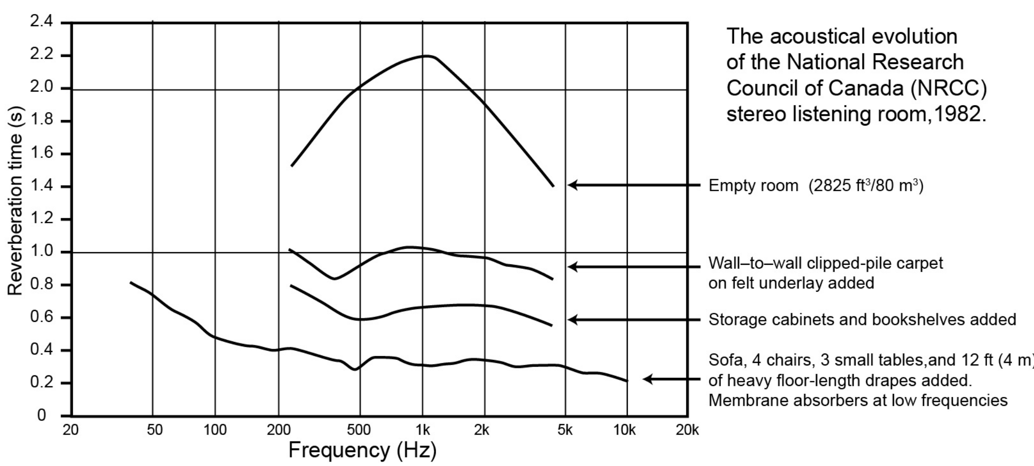


Figure 4. The evolution of a normal domestic listening room.

Let us start with the acoustical performance of some materials that could be found in normal living spaces. Figure 4 shows the changes to RT as various materials and objects were introduced into an empty room. It is evident that wall-to-wall carpet and felt underlay had a huge effect on RT. Adding storage units and bookshelves contributed considerable scattering, which made the carpet work harder as sounds were redirected into it. The books added some absorption as well. Completing the furnishing of the room with upholstered furniture, tables, and some drapes brought the RT into what then, and now, is considered to be a desirable range for quality listening experiences. The membrane absorbers operated below 100 Hz, and were necessary only because the room was originally a laboratory space with highly reflective, stiff and massive masonry boundaries. Wood or metal frame + drywall rooms would probably not need them.

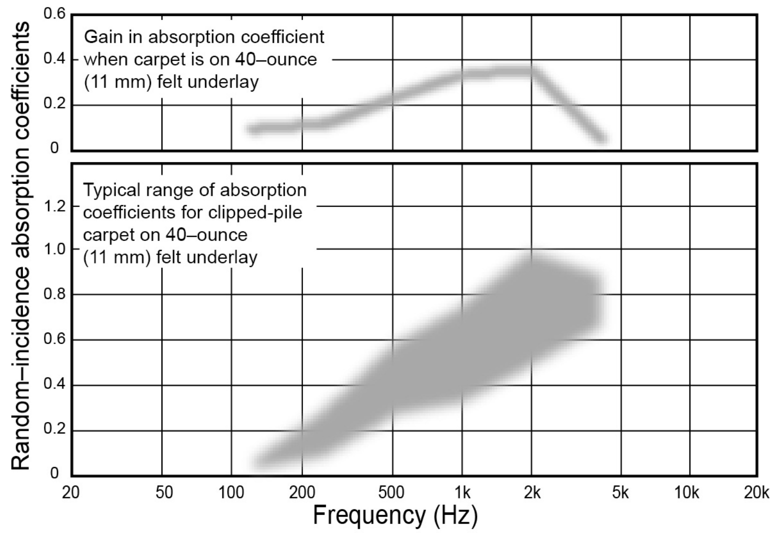


Figure 5. At the top is shown the isolated influence of a good carpet underlay—40 ounce/sq. yd (1.4 kg/m^2) hair felt which is typically about 0.43 inch (11 mm) thick. The same thickness of common rubber, plastic or foam cannot deliver this level of acoustical performance. There may be other materials that are comparably good, but check for acoustical measurements. The shaded area in the bottom curve combines this underlay with different kinds of high quality clipped-pile carpets with porous backing—the sound must be able to penetrate into the felt underlay.

Figure 4 showed the profound effect that wall-to-wall carpet had on the reverberation time of an empty room. Figure 5 shows why; a good carpet on a good underlay is an effective acoustical absorber and it is easy to justify a lot of it in a room. As a resistive absorber, it is most effective at middle and high frequencies, which is why it is important to maximize the effective thickness by using an acoustically useful (as opposed to just comfortable under foot) underlay. Because it is all on one surface, we need sound scattering/diffusing objects and surfaces to tame flutter echoes between walls, and to redirect sound into the floor. Again, Figure 4 shows that adding non-absorbing, but scattering "stuff" to the room substantially reduced the reverberation time. So, furniture is important. The acoustical properties of carpets vary substantially, and this is not a published property of the product. The basic rule is that if absorption is required, avoid the looped-pile rubber-backed industrial "indoor/outdoor" style of product.

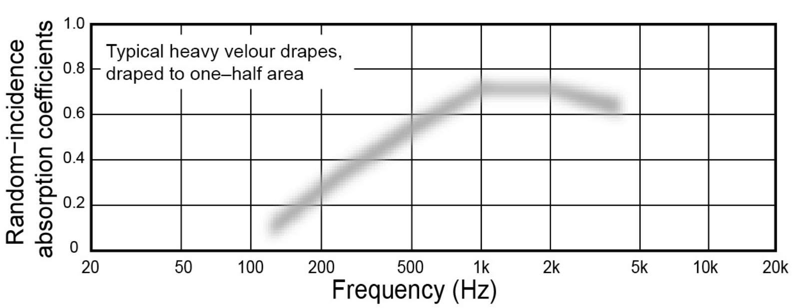


Figure 6. A fuzzy curve showing approximate absorption properties of heavy velour drapes, draped to one-half of their flat area. The drapes should be hung on a track located 4–6 inches (10–15 cm) from the wall in order to ensure some low-frequency absorption.

As shown in Figure 6 drapes can also be effective sound absorbers if they are of the right kind. Heavy velour, the heavier the better, is probably the most popular choice because in its draped form it constitutes a resistive absorber of significant thickness. Other heavy fabrics with the right porosity/flow resistance also work well, but lightweight, open-weave fabrics (easy to blow air through) are useful only for decorative purposes—the best absorbers impede airflow but do not block it.

Upholstered seating performs two important functions, absorption and scattering. Chairs, especially of the home-theater variety, are substantial obstacles in the middle of small rooms and some of the sound falling on them is absorbed, and some is redirected by reflection and scattering elsewhere in the room. They interact with a significant fraction of the direct sound radiated from the loudspeaker, and they effectively absorb sound over a large frequency range, including lower frequencies (see Figure 10.5, p. 290).

The most natural place to look for absorption to help with low-frequency room resonances is in the room itself. Significant absorption is available from normal gypsum-board-on-stud constructions; when you can feel bass in the walls and floor, membrane absorption is taking place. The absorption coefficients are not very high, but there is a lot of wall and ceiling area. A single layer of drywall is most effective with absorption diminishing as layers are added—as is often done for sound isolation (see the PowerPoint tutorial on this website). A second consideration is that there will always be some of the absorption in the correct location to provide damping for all room modes (must be at high-pressure points in the standing wave patterns). If sound isolation is not a consideration, there is little to ponder—use a single layer of drywall. If sound isolation is important, another wall, outside of this one will be needed (room within a room design). Remember, in multiple-wall structures, the distance between the separate wall surfaces is a prime determinant of sound attenuation (along with mass), so be sure to allow several inches between the outer surface of the inner “room” and the inner surface of the outer “room,” or omit the inner surface entirely, adding more layers of gypsum board to the outside.

Random-incidence absorption coefficients tell part of the story, but what really matters is the effect on resonances in real rooms. A few years ago, an opportunity presented itself to find out. A listening room had been built, and the friendly builder thought he was doing us a favor by adding an extra layer of gypsum board on the interior surface. The “favor” was discovered too late, and the room was used in that condition for a couple of years during which it acquired a reputation for having somewhat boomy bass and upper-bass/lower midrange coloration. The evidence was in measurements. Eventually, we had the interior of the room stripped and the originally intended single layer of gypsum board installed. Before and after measurements are shown in Figure 7. The benefits were measurable and audible. The surprise was how high in frequency the effects extended. The resonance peaks are significantly attenuated but some dips are more responsive than others. This is because the dips are subtractive (destructive interference) effects caused by two sounds of opposite polarity interacting. After the absorptive attenuation both sounds still exist, but perhaps in different magnitudes, and still subtract. These very narrow-band phenomena are not the audible problems that resonant peaks are.

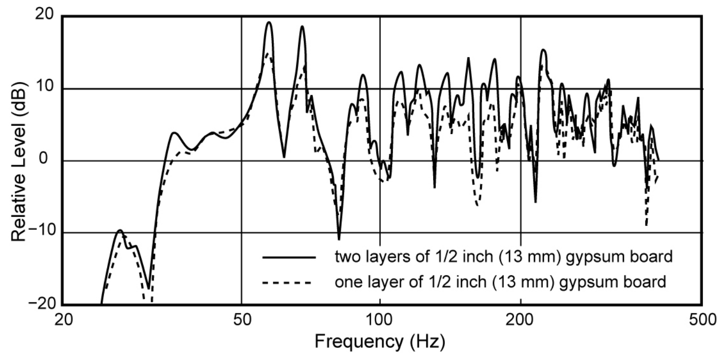


Figure 7. Measurements made with a subwoofer placed in a front corner, and a microphone located at the prime listening location in a 3000-ft^3 (85-m^3) listening room. The solid curve shows results with two layers of gypsum board and the dashed curve shows results for a single layer of gypsum board mounted normally on 3.5-inch (90-mm) wooden studs. It is clear that the single layer provides a substantial increase in acoustical damping at frequencies below about 200 Hz. It also shows that the overall sound level has been reduced as some of the bass energy has been "trapped"—the loudspeakers will have to work harder to create the same sound levels.

Rooms have been built with sand-filled walls, having heavy plywood and multiple layers of gypsum board on the exterior surfaces for strength. In the example that I encountered the stated purpose was to keep the bass in, to make it "tight." It succeeded in keeping it in, excessively so, and the result was a room with enormous undamped bass resonances. Far from being "tight," bass boomed mercilessly. The solution required the addition of costly and bulky low-frequency absorbers to return the room to a state that could have been achieved with normal household construction materials and methods. In fairness, the sound transmission loss through the massive wall was considerable but, ironically, in this industrial building it was not a requirement.

If steel studs are used, extra precautions are needed to avoid buzzes and rattles. At the very least it is necessary to substantially increase the number of attachment screws, and preferably to run a bead of acoustical caulk down each stud before the gypsum board is applied. This done, such walls are eminently satisfactory, and there is the additional advantage that lightweight steel studs offer better sound isolation than wood, because they flex.

8.1 Sound Absorbers

The normal porous resistive absorbers are fibrous tangles—of which glass fiber, mineral wool and acoustical (leached) foam are the most common (see Section 4.10.3, p. 102). Glass fiber and mineral wool were created primarily because of their thermal insulation properties. They are therefore available in many forms, from soft, flexible batts or rolls, to rigid boards in which the fibers are compressed with an adhesive. The latter are available in different thicknesses and densities rated in pounds per cubic foot (pcf) and kg/m^3. Measured in the traditional reverberation chamber fashion, which typically covers the 125 through 4 kHz octave bands, there is little difference in the measured absorption coefficients for the same thickness of any of the materials, soft or hard fibrous materials or slab foam.

All measurements have tolerances, and these are no exception. Even though the measurement facilities may have been certified to meet industry standards there are differences among them. Figure 8 shows the accepted variances among measurements, and it can be seen that it embraces data on five different 1-inch (25-mm) fibrous materials. Some of the touted "superiorities" of different materials may simply be evidence of normal measurement variations among different testing facilities. However, because directional absorption performance is what truly matters in small rooms, none of this is consequential.

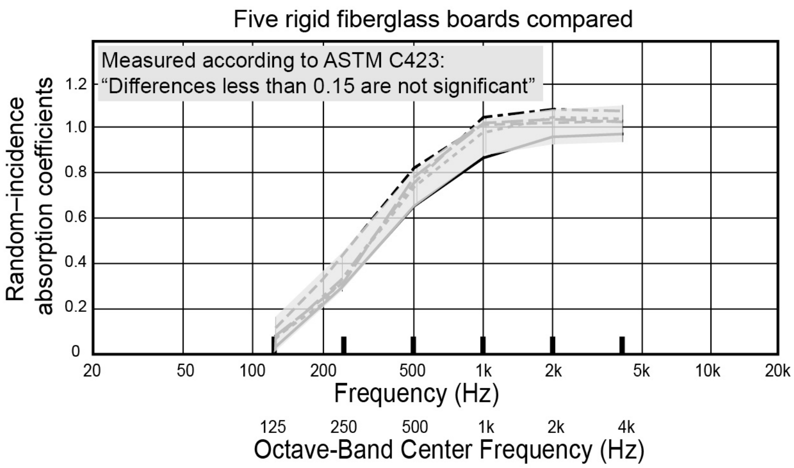


Figure 8. The gray area shows the industry standard tolerance on the precision of reverberation chamber measurements of random-incidence absorption coefficients. It embraces five different 1-inch (25-mm) materials of differing manufacture and densities.

Acoustical absorbers are most often made from rigid fiberglass board, or the like, covered with fabric. They are convenient to install, and decorative. However, the reality is that the high densities that are necessary to create structural strength do essentially nothing to alter the acoustical absorption properties of the material. If cost saving is a factor, much lower density materials work perfectly well, including the readily available batts used for household insulation. In general, lower densities are preferred. Be careful to use acoustically transparent fabric covering—it should be easy to blow through, and speckles of light may be visible through it. A fire-rated fabric is advised, if not a legal requirement.

Random-incidence sound absorption coefficients for different thicknesses of material are widely available on the Internet. They show improved performance at lower frequencies as the depth of the fibrous material increases. At around 3 inches (75 mm) the materials are usefully absorptive down to 200–300 Hz. It is worth considering the cost savings involved in using a thin high-density board for the external layer of a wall treatment, and filling a space behind it with low-density, inexpensive fluff. Any depth can be achieved in this manner. The empty space above a dropped T-bar ceiling can turn thin ceiling panels (which must be of the rigid fiberglass board type) into quite effective broadband absorbers. Adding loose fluff, such as attic insulation, in the cavity above improves performance. Obviously, the installer must take great care in assembling the T-bar metal support system to avoid rattles and buzzes, but it can be done, and it works. Such a scheme provides considerable low-frequency absorption, which normally is difficult to achieve. The entire ceiling becomes an "invisible" broadband absorber and bass trap.

In listening rooms the sound fields are not diffuse or random incidence, as is assumed for the published specifications. Instead, they are strongly directional, as is discussed in detail in Section 10.2 (pp. 286–291). Consequently, the random-incidence data are not directly applicable. If one considers only the side wall reflection, the incident sound arrives from about 45° and the frequency dependent absorption is *very* different from that which would be anticipated from random-incidence data. As described in Section 7.3 (p. 167) and Figure 7.10 (p. 174) the real effect of commonly used materials is simply to roll off the high frequencies. The reflection is not eliminated, but the perceived sound quality of the (possibly good) loudspeaker is negatively influenced, and the spatial illusions modified in an unpredictable manner. It is no surprise that there are differing opinions about the effects of side wall reflections.

At low frequencies, as discussed earlier, some conventional wall construction methods contribute useful amounts of absorption. However, this may not be enough, and the next resort is to employ membrane/diaphragmatic absorbers—what are colloquially called "bass traps." These can take several forms, useful for different applications, but a serious caution is warranted. Some are really "upper-bass traps" and are not very effective at the below 100 Hz frequencies where many serious room booms happen. Look for credible specifications of performance. Many of these devices are broadband, meaning that they may address a problematic room mode, but also remove energy at frequencies that are not problems. There are instances where some amount of tuning of the frequencies being absorbed is advantageous.

If the multi-sub solutions discussed in Chapter 8 are employed, little or no additional low-frequency absorption may be required. However, if starting with a clean sheet of paper, it is always advantageous to incorporate some low-frequency absorption—it simply makes everything else work better.

8.2 Sound Diffusing/Scattering Devices

This topic is well covered in the book, in Section 4.10.4 (p. 103), Section 7.3.2 (p. 169) and Section 10.2 (pp. 286–291). The shallow devices are effective in alleviating "hand clap" flutter echoes, but many of these are little more than placebos. We are not entertained by mid-to-high-frequency hand claps. In reality much wider bandwidth, inevitably deeper/thicker devices are needed to be truly effective. The frequency dependence of their scattering abilities is fundamentally important.

8.3 Flutter Echo

These are the "zings" of sound ricocheting between parallel reflecting surfaces in response to an impulsive sound like a handclap. They are easily heard by the person clapping the hands but may or may not be as consequential to persons elsewhere in the room.

Problematic flutter echoes happen, but the ones that matter most are those excited by the loudspeakers and heard by the listeners. So, the likelihood of a problem increases with the number of loudspeakers in the room. The critical test is to have a person stand by each of the loudspeakers in turn, clap hands, while a second person listens from the seating area to see if there are audible flutters. If so they are easily treated by absorption or scattering, often not much of either. Although thin 1–2-inch absorbers and diffusers will eliminate handclap flutter, it must be remembered that they generate mainly high frequency sound. It is highly likely that wider bandwidth sounds from the loudspeaker will also “flutter,” in which case thicker materials are advised.

This is easy to investigate and cure in stereo, not very complex in 5.1 or 7.1 systems, but with immersive multichannel systems more attention to detail may be required. This explains the elaborate treatment patterns suggested in Figure 2.

9 Conclusion

The most important thing to remember is that if one begins with well-designed loudspeakers (as described in Chapters 5 and 12), above the transition frequency of about 300 to 500 Hz, the difficult work is substantially done. This may sound like a commercial because I am still associated with a manufacturer, but the fact is that there are several brands that have learned the relevant science and applied it to their products. The necessary information is all published and in the public domain. Many, though, have not learned it, don’t believe it, or simply choose to go their own way, and that, combined with an absence of comprehensive and trustworthy measurement data, is the problem for consumers and professionals. Folklore and opinions are widely available; facts are scarce.

Because humans adapt to many acoustical aspects of rooms there may be little or nothing to do in a well-furnished living space or a suitably treated dedicated space. If the RT is below about 0.5 s, speech intelligibility and music articulation should be satisfactory, and if there are no overly prominent or long-delayed reflections, the fundamentals are in place.

Effort and money can then be directed towards delivering high quality bass, which means attending to the room resonance/standing-wave issues (Chapter 8) and the adjacent boundary issues (Chapter 9). The importance of deep, clean non-resonant bass cannot be overstated. It is *not* a matter of the quantity of bass, it is the uniformity and bandwidth. All of the musical notes need to be there in the right proportions, down to and including organ pedal notes and synthesizer effects. Percussion should be tight, non-resonant. When you get it, you realize what has been missing, and, interestingly enough, these qualities can be appreciated at moderate volume levels. That said, occasional whole-body experiences are thrilling too.

This introduces the topic of subwoofer performance below 20 Hz, which is commonly regarded as the lowest frequency with recognizable musical pitch, and even that is a stretch. What humans experience at these subsonic frequencies is more feeling than hearing. There is almost nothing in normal music or movies that deliberately addresses this part of the spectrum (some organ pedals and synth tones are exceptions). Nevertheless, there are examples of subsonic content. It is probable that only in the rarest of instances did the mixers experience it in a control room or dubbing stage. If it is noticeable, my own observations are that the “whole body” responses are mostly associated with impulsive sound effects in movies and some modern music selections. Occasionally one is aware of subsonic HVAC in the recording environment, which is evidence of a control-room monitoring system of insufficient bandwidth. I also recall from decades past, reflex-loaded woofer cones fluttering wildly to very low frequency surface irregularities and tone arm resonances in LPs, and modulating higher frequencies. Subsonic reproduction isn't a bad idea, but it is a double-edged sword.

Always keep in mind that hearing involves a cognitive component. Humans are highly adaptable, modifying our perceptions to make us more comfortable. An excellent example of that is a recent study that found that listeners preferred a floor reflection—it sounded more "natural" (Section 7.4.7, p. 193). Clearly, throughout human evolution there has always been a reflecting surface under our feet and its contribution to what we hear is "expected."

This example teaches us that not everything we can measure is a target for acoustical treatment. But, there are limits to adaptation, and this is where dedicated corrections may be needed. Perfection, if indeed we could define it, is not a requirement. We adapt to our circumstances; we "listen through" rooms to a great extent. Acoustics is an important topic to be sure, but never underestimate the importance of *psycho*acoustics.

Finally, some things that should be obvious need to be emphasized:

- Rooms are three-dimensional physical entities, with boundaries and contents that absorb, reflect and scatter sound.
- Loudspeakers radiate sounds in all directions—different sounds in different directions.
- As a result, sounds arriving at the listening position differ in spectrum, timing and incident direction.
- The process involving two ears and a brain is extremely complex and analytical, yielding multidimensional perceptions of sound quality, direction and spatial impressions. Human listeners respond differently to reflected sounds arriving at different times from different directions. Omnidirectional microphones are "deaf" to directional cues, and steady-state measurements also ignore timing differences.
- An omnidirectional microphone and an analyzer cannot describe this entire process. The notion that "room equalization/calibration" can transform unknown rooms and unknown loudspeakers into perfect combinations is not credible.

Music and movies can be enjoyed through less than perfect systems, but when experienced through a broadband spectrally neutral one, the improvement is not subtle and the effects are highly pleasurable. The world needs more of them. With such a system, it becomes clear that the variations in programs are a significant weak link. The situation seems to be improving, but, especially when listening to older programs, it is obvious that consumers and professionals have not always been listening to the same kinds of sounds. It is time for us all to be "on the same page."

Last updated on October 15, 2017

Designing Home Theaters and Listening Rooms:
Part 2—Loudspeaker Selection, Placement, and Calibration

by

Floyd E. Toole, PhD

Outline

The 500-page book *Sound Reproduction—the Acoustics and Psychoacoustics of Loudspeakers and Rooms*, by Floyd E. Toole, Focal Press, 2017 has in-depth discussion and data on many topics. In this much-simplified guide readers will be provided with some additional information, and directed to parts of the book using figure and section numbers and page numbers for more explanation.

Other documents on this website will also be referred to. The website will change with time, so updates to this guide will follow; check the edition date at the end.

1 Introduction

There are three basic arrangements of loudspeakers to be considered at the present time:

- Stereo
- 5.1 and 7.1 surround sound
- Surround sound enhanced by elevated loudspeakers: immersive (3D) audio.

Stereo has been with us for about 60 years, and it remains the default format for music. Movies were a motivating factor for stereo, and movies have driven the desire for surround sound and now immersive sound. There are examples of multichannel music, and some are impressively good, but the impracticality of maintaining inventories of different physical formats has been a major cause of music still residing in two channels. Current streaming capabilities and the flexibility of downloadable playback algorithms offer hope for a future with an expanded repertoire of multichannel music pleasures.

Multichannel soundtracks accompany many music videos, they are compulsory in movies and some productions for TV are incorporating more elaborate audio. Upmixers can manipulate stereo music into surround or even immersive versions. Because the creators of stereo music did not anticipate such manipulations, not all of it responds favorably to upmixing. However, the author has found that a moderated application of upmixing provides a very pleasant increase in "envelopment," tending to include the listener in the performance space. This is quite successful in the classical repertoire, and surprisingly often in the popular/jazz domain. The "stereo" button is always there. I rarely use it.

Finding a pleasing upmixer is the challenge. Some popular ones are (for me) overly aggressive in enhancing the center channel and/or directing sound to the surround channels. However, modern receivers and especially powerful surround processors provide many customizable adjustments to rebalance amplitudes and delays for fronts and surrounds. Some experimentation of this kind is actually educational in that it quickly reveals just how different stereo mixes can be.

The point of this discussion is merely to point out that state-of-the-art stereo is available in multichannel audio systems but the reverse is not true. Therefore, in designing listening rooms, it is advisable to design a system that looks to the future, and has, or can have, multichannel capabilities. At the very least this means running wires through the structure for additional loudspeakers in some key locations. Some builders of new homes currently do this, but lacking good guidance, they sometimes get the locations wrong.

Defining those locations is one objective of this presentation. The L, C and R loudspeakers are well positioned to deliver excellent sound because they are, or should be, aimed at all listeners in the room. I still see photos of home theaters with all three front loudspeakers aiming straight out into the room. Why?

Side, rear and overhead loudspeakers can be more challenging. The reason is that, depending on how the loudspeakers are designed, positioned and mounted, the direct sound arriving at a listener might have left the loudspeaker at a considerably off axis angle. The important direct sound is degraded. Equalization to correct it will distort all other sounds leaving the loudspeaker, which negates one of the engineering objectives of good sounding loudspeakers. I often joke about a simple in-ceiling loudspeaker being optimally positioned to entertain the dog lying on the floor under it, not the humans in the room.

This means that some thought must be given to the kind of loudspeakers used in the surround/immersive systems if listeners are to have the highest quality experience. This includes constraints imposed by the dwelling itself and by the customers, many of whom want "invisible" loudspeakers. Obviously, apartment dwellers cannot use in-wall/ceiling designs, so on-wall or bracket mounted small loudspeakers are the options. Listeners should be situated on axis or within a reasonable off-axis listening window of the in-/on-wall/ceiling loudspeakers for sound quality to be maximized. Some scaled room layout drawings and a protractor are required (or the computer graphic equivalent).

2 Basic Stereo and Multichannel Floor Plans

The principal requirement for stereo is two identical loudspeakers at equal distances from the listener (it is designed for only one listener) and located at about ± 30°, although smaller soundstages are common.

Chapter 15 provides a lot of detailed information on optimizing multichannel layouts. The following is just a sample.

The first iteration of surround sound employed four channels, L, C, R and one channel for the left and right surrounds. Fortunately, it was not long before the situation improved, first with decorrelated left/right side channels to improve the spatial impression, and then with five digital discrete channels: 5.1. The ".1" was for the subwoofer. Five discrete channels allow the recording engineer to control the delivery of directional and/or spatial impressions.

In the beginning, following the guidance of the defunct "quadraphonics," people often interpreted the surround channels as "rear" channels and placed the loudspeakers behind them. As seen in Figure 1(a & b), the surround channels should be positioned mostly to the sides, to deliver good envelopment, and slightly behind the listener to create a slight rearward tendency so that flyovers can be effective. The worst possible (but all too common) arrangement is to have the rear loudspeakers at the same mirrored angles as the front L and R loudspeakers (see Figure 15.3, p. 407)—in terms of generating spaciousness, it is no better than stereo, although flyovers would be very persuasive.

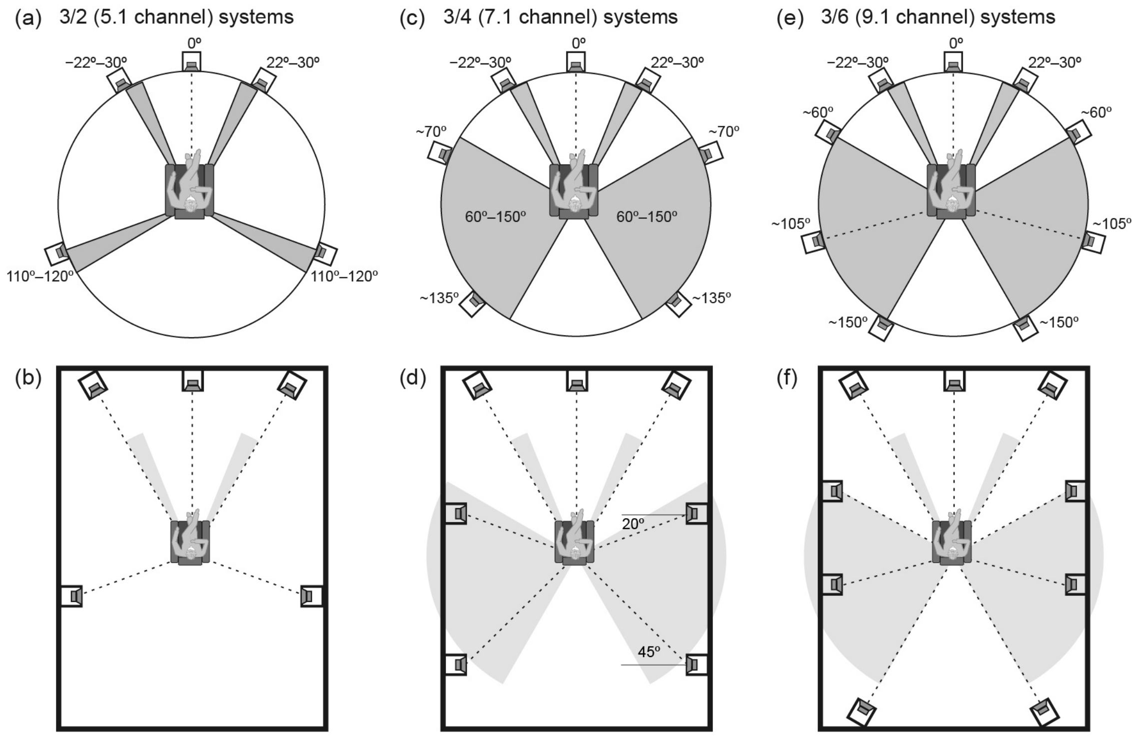
It was not long before multichannel evolved into 7.1. This generally has been interpreted as 5.1 with two more channels added at the rear. However, if the desire is to optimize the spatial display, it is advisable to move the side loudspeakers forward for maximum spatial effect, and let the rear loudspeakers deliver impressive rear localizations and flyovers. This is shown in Figure 1 (c and d). If possible, this is the preferred layout.

If space is available, and the electronics can support the additional loudspeakers, Figure 1 (e and f) shows a 9.1 channel arrangement. If the −60° and −105° loudspeakers are driven by the same signal, as if they were a combined side channel, the −105° pair should be delayed by about 10 ms to add sufficient decorrelation to avoid interference effects (comb filtering). If they are used in immersive formats they are separate program channels and no special treatment is needed. Observant readers may have noticed that in (e and f) the rear loudspeakers mirror the front L and R, but here it is of no consequence because there are four side loudspeakers better positioned to deliver spatial effects.

Circular arrangements use a lot of space, are incompatible with most domestic rooms and, consequently, are rare. Differing distances in the common rectangular arrangements need to be compensated for using delays in the setup routine. The side wall loudspeakers have all been illustrated as on-wall types. They could have been in-wall designs. In either case Figure 1 (d) shows that the prime listener is 45° off the design axis of the rear loudspeakers. This is not likely to deliver a flat direct sound to the prime listener, and equalization is not a desirable solution. Elevation loudspeakers at ceiling height sometimes present even greater off-axis problems. There are two options: use bracket-mounted loudspeakers and aim them at the listener, as in

Figure 1 (f); or select a loudspeaker that has good performance at that off-axis angle. The front side loudspeakers are within a reasonable listening window for this listener.

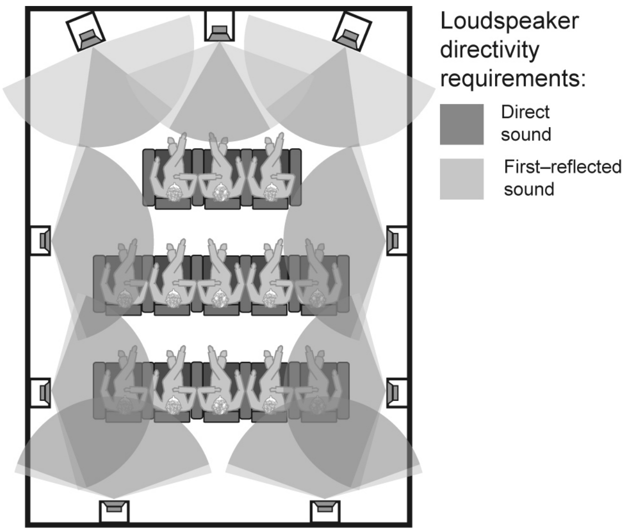
Figure 1. Suggested arrangements for common multichannel formats.



2.1 Surround Loudspeaker Directivity Requirements

Home theaters have multiple listeners, sometimes several rows of them. If it is wished to deliver pristine direct sounds to all of them, some thought is required. Figure 2 shows an extreme example of this, indicating that without the capability of aiming the side and rear loudspeakers, very large horizontal dispersions are required from typical in- or on-wall units. Delivering an intact direct sound is especially important in immersive movies because it is the direct sound that determines the perceived direction of the sound or sound effect. Perhaps more importantly, the direct sound is the prime factor in determining perceived sound quality.

For this reason, it is *always* good to start by considering the use of high quality surround and immersive loudspeakers that can be aimed at the prime listening position. That way the direct sound will be good, and early-reflected versions of it will be as they should be. Attempting to equalize the off-axis sound of an inappropriately aimed loudspeaker to be flat may or may not be an improvement, but it cannot deliver the best possible sound quality. Perhaps there should be a category of home theater in which loudspeakers are unashamedly visible, as a symbol of having pride in delivering to listeners the best possible sound from loudspeakers that provide at least half of the pleasure in movies and all of the pleasure in music. I fear it is a losing battle . . .



Horizontal dispersion requirements for the loudspeakers in this crowded room with a large audience:
LF, RF: ±30º for the direct sound
±87º for first–reflected sounds
Center: ±30º for the direct sound
±70º for first–reflected sounds
Surrounds: ±70º for direct sounds
±87º for first–reflected sounds
Loudspeakers are manufactured with symmetrical left/right dispersion whether it is needed in a particular location or not.

Figure 2. A summary of the horizontal-plane angular dispersions required of the loudspeakers to deliver direct sounds of comparable quality and level to all listeners and to deliver sounds to the wall surfaces from which the first reflections occur. The propagation loss due to the inverse-square law will inevitably cause differences in level at different distances. The criterion of excellence for direct sounds (the darker shaded angular range) is that they should all be as similar as possible to the on-axis performance of the loudspeaker. This is obviously a challenge for the surround loudspeakers because of the very large, almost 180° dispersion required of these units. Here the surround loudspeakers are all non-steerable in-wall or on-wall designs. (This is Figure 15.9, p. 421)

In Figure 2 it can be seen that conventional forward-firing loudspeakers will be beyond their normal listening windows (±30° horizontal) if they are mounted flush with the wall. Bidirectional in-phase (bipole) designs will work well, but dipole (bidirectional out-of-phase) designs should be avoided (see Section 15.8.3, p. 422).

If the room can accommodate the extra depth, in-wall loudspeakers can be built into tilted supporting structures, or appropriate aiming of bracket mounted conventional loudspeakers will work. The latter has the advantage that if they are from the same product line, they are likely to be a good timbral match to the L, C and R loudspeakers. This is an advantage.

As discussed in Section 9.3, p. 269, mounting “bookshelf” loudspeakers on or near a wall has an adjacent-boundary effect (Chapter 9). There will be a narrow acoustical interference dip in the direct sound (Figure 9.8, p. 271) but only a shallow, broad dip in the overall spatially averaged room sound (Figure 9.10, p. 273). It may or may not be an audible problem because there are other room effects in that frequency range that can compensate for, mask, or confuse what a listener hears. Equalization will be necessary to compensate for the boundary-induced bass boost, but it may be unwise to try to fill the narrow-band interference dip in the direct sound (a non-minimum-phase effect) that changes with microphone/listener position. The shallow broadband dip will be part of steady-state measurements in the room, appearing as an energy deficiency that can be compensated for using low-Q filters (Section 9.2.1, p. 267).

2.2 Elevated Loudspeakers for Immersive Formats

I personally am currently facing a decision of how many additional loudspeakers to incorporate into my under-revision home theater. In many processors and receivers the choices are limited, and there is little or no flexibility. Four ceiling-height loudspeakers seems to be a popular choice: two towards the front and two towards the back of the room. However, if reproducing all immersive formats is desired, six elevated loudspeakers, three at the front, one "voice of god" in the center and two at the rear, are probably advisable. A quick Internet search will reveal what various manufacturers are offering.

The underlying problem is that there are three main providers of immersive audio: Dolby Atmos, DTS:X and Auro-3D. Each one employs different recommended arrangements of immersive loudspeakers, reaching large numbers in the most elaborate configurations. These may be necessary in large cinema installations, but in home theaters they are simply impractical. Cinemas also have budgets to consider, so many of those install depopulated versions. In the professional domain the matter is not entirely settled if a cinema wishes to offer movies mixed in different immersive algorithms.

Some processors claim that they have the ability to "virtually" reconfigure how signals are sent to whatever loudspeaker arrangement is decided on. Up to a point this is possible in the object-oriented formats, but even there the effect is optimized for the prime seat; it cannot work for all seats. When sound effects are shared among multiple loudspeakers, which is the norm in object-oriented formats, acoustical interference among them affects sound quality, and it will be affected differently for different immersive loudspeakers and loudspeaker configurations. There are a few fortunate facts:

- Many (most?) of the steered immersive effects are sound effects in which "fidelity" in the normal sense does not apply.
- Many of the steered immersive effects are not lost on audiences if they appear from the "wrong" locations, and
- Humans are very poor at localizing sounds in the vertical axis (our ears are in the horizontal plane). In the excitement of a good drama, high precision is not a requirement.

Somewhere there is a compromise between what is possible and what is necessary. At present, there are forces at play on both sides. Obviously, the directivity issues discussed in the previous section apply to all immersive loudspeakers as well.

2.3 Seating Issues in Small Rooms

Rear wall reflections:

Many home theaters exist in rooms so small that the seating is close to or against the rear wall. This is a problem for three reasons:

- Sound from the front loudspeakers reflects off the wall behind the listeners and distorts both the timbre and soundstage (see Section 9.6, p.279). This is especially damaging to stereo phantom images.
- Standing waves always have high-sound-pressure regions adjacent to walls, meaning that there is a high probability of those listeners hearing excessive bass. Mode attenuation schemes can be advantageous (Section 8.2, p. 224).
- It is not possible to position loudspeakers behind the listeners so some directional effects are not possible.

The reflected sound can be attenuated by placing an effective absorber on the rear wall. At least 6 inches (150 mm) of fibrous absorber behind an acoustically transparent fabric is recommended, and it should extend well above and below ear level. Do ***not*** put a diffuser close behind a listener. This is done in some recording control rooms following the style of one end live and the other end dead. Some control rooms have large areas of diffusers. It is easy to determine if there is a problem. Play monophonic pink noise, belly up to the center of the console and see if you hear a well-defined phantom image floating midway between L and R loudspeakers. I have experienced control rooms in which there is ***no*** phantom center image, just a blurry wall of sound. The reason: strong uncorrelated sounds arriving at the ears that disrupt the direct sounds from the front loudspeakers. This problem is obviously greatest for listeners closest to the diffusers.

Off-center listening:

A second common problem has to do with listeners who are seated well off-center, where they can experience either or both of two problems:

1. Those who are too close to the side loudspeakers are able to localize them in situations when no specific sounds should be coming from them. This is distracting.
2. Envelopment is best perceived when the sounds at both ears are similar in level. This happens midway between the side loudspeakers, but at seating locations closer to the walls the effect is rapidly diminished. See Section 14.2.3, p. 391.

The localization problem has no really satisfactory solution. Attenuating the high frequencies helps because the most localizable component of sound is high frequency transients. This is essentially what was accomplished by bidirectional out-of-phase (dipole) surround loudspeakers, but only for one row of listeners. The problem is that this degrades sound quality. Elevating the side loudspeakers and arranging for the tweeters to fire over the heads of the nearest listeners is a way to achieve some of this effect in a relatively harmless way. Designating the side seats for non-critical listeners is perhaps the only practical solution.

The envelopment issue is also one where there is no easy solution with conventional loudspeakers, as described in Figure 14.4, p. 394. Sitting close to the center of the room is advised. Some premium cinemas and home theaters use CBT designs from JBL Pro to deliver more uniform side-to-side sound levels. These can approximate the performance described in Figure 14.4 (d), thereby expanding the desirable listening area. They also address problem 1, without changing the sound spectrum.

Both of these problems are also experienced in cinemas, but the larger distances allow more listeners to be in good seats.

2.4 Center Channel Issues

In movies and television programs the center channel does most of the work. It is arguably the most important loudspeaker in a home theater. But it is frequently a problem to find a place to put it. Virtually all manufacturers offer horizontal versions of left and right loudspeakers for center use. Timbre matching can be excellent.

Still, the matter of where to put the center channel is a problem that won't go away. With direct view displays the common options are: above or below. Whichever option is chosen, all listeners should have line-of-sight contact with the center channel. It is fortunate that:

- The positioning of the ears means that human localization in the vertical plane is not at all precise, so the fact that the sound is not coming from the mouths of the actors or newscasters is less obvious.
- We are very susceptible to the ventriloquism effect in which we are more attracted to the moving lips, flashing gun, or slamming door, than to the actual source of the sound. In many movies most of the on-screen sounds emerge from the center channel and that includes sounds other than dialog, yet audiences at home or in cinemas don't seem to notice the discrepancy even in the horizontal plane where we are very sensitive to directional cues. Visual cues dominate.
- Humans adapt. Even if one may notice a directional discrepancy upon first exposure, most listeners quickly adapt, compensating for the situation.

The possibility of placing center loudspeakers both above and below the display is sometimes suggested. It is not recommended. As explained in Section 15.12.1, p. 430, elevation cues are at very high frequencies, so if one wishes to attempt to elevate the apparent location of a below-screen loudspeaker all that should be needed is a tweeter radiating sound above 5–6 kHz or so located above the screen. It would make an interesting experiment.

With front projection displays there is the option of placing loudspeakers behind a perforated vinyl screen or woven fabric screen, in which case they can be identical loudspeakers. Screen loss is significant with perforated screens but simple equalization can compensate (Section 10.7, p. 300). Nevertheless, fastidious listeners who use their home theater systems for stereo listening often place the L and R loudspeakers outside the frame of the screen.

If a horizontal center loudspeaker is the selected option, select a three-way system with a midrange and tweeter vertically arranged in the center of the system. The horizontal dispersion is significantly more uniform than the simple tweeter in the middle systems. Figure 3 illustrates the difference in horizontal directivity.

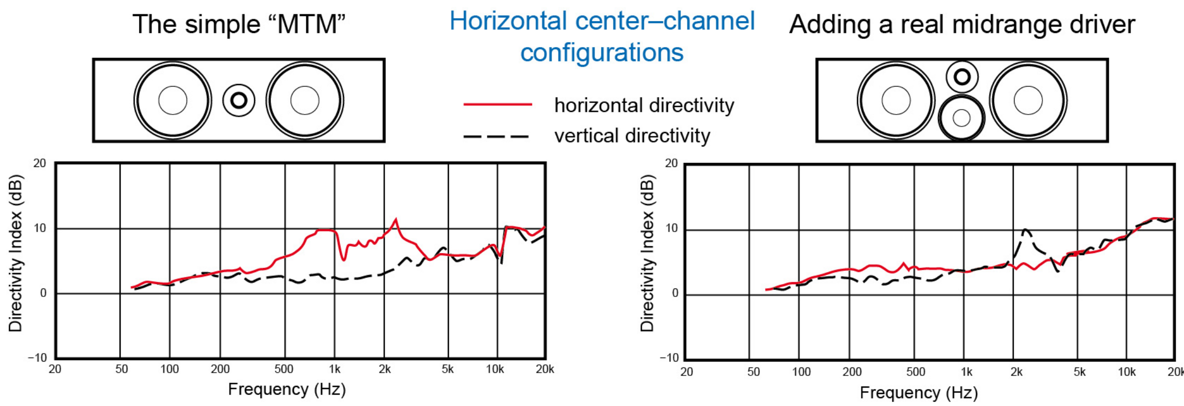


Figure 3.

3 Subwoofer Options

Nobody doubts the importance of bass in either music or movies. Yet, in most systems it is treated in unsophisticated ways—a bigger, "better" subwoofer is the accepted solution. This is simply wrong. Research has shown that bass extension and smoothness accounts for about 30% of our overall opinion of sound quality (Section 5.7, p. 135). This is certainly affected by the choice of subwoofer(s), but in all cases the room dominates what we hear. Until it is tamed truly good bass is not possible. Some of the methods of controlling annoying room resonances also reduce seat-to-seat variations, as well as being beneficial to the quantity of bass that is delivered.

The problem in small rooms is room resonances and the associated standing waves (Chapter 8, p. 215). The traditional solution has been to add bass traps to damp these resonances, and to involve room equalization to smooth the frequency responses. Recent research has shown that the much-discussed time-domain ringing of these resonances (seen in picturesque waterfall diagrams) is not the primary audible characteristic (Section 8.3, p. 255), but it is associated with the peak in the frequency response that most definitely is. Attenuating resonant peaks in low-frequency room curves is the primary objective. Attempting to fill narrow notches or nulls is likely to make things worse. Knowledgeable use of manual equalizing filters is a wise approach, but today automated equalization is widely used. Not all algorithms are programmed with the discipline to ignore narrow destructive interference dips, which helps to explain differences in opinions about these schemes.

Damping the resonances with low-frequency absorption is the traditional solution. To the extent that it can be done using membrane absorption in the room boundaries there is no visual effect on room décor. However, if that is not sufficient, adding bass traps is the option, but they are large and unattractive in normal domestic spaces. In professional audio and in custom home theaters it is possible to either accept the visual compromises imposed by bass traps (possibly even being proud of them), or find ways to disguise them.

The real problem is standing waves, which cause bass to boom at certain frequencies, to be absent at others, and to be different in different locations in the room. Equalizing a single subwoofer can improve the sound at the microphone location, but nowhere else. A single subwoofer has no ability to reduce seat-to-seat variations or remove peaks and nulls. Multiple subwoofers can do both, presenting an opportunity to attenuate resonances, alleviating the associated pesky peaks and nulls in the standing waves. Chapter 8, p. 215 describes the options for reducing the detrimental effects of room resonances on bass. Some solutions work only in rectangular rooms, reducing the number of active resonances, and creating areas of rooms within which seat-to-seat variations are reduced. In the most advanced solutions, the resonances and seat-to-seat variations are essentially absent. All of the most effective solutions involve multiple subwoofers and these are well explained in Chapter 8.

It needs to be made clear that multiple subwoofer methods of attenuating room resonances *only* do so when the sound reproducing system is active and they *only* affect sounds being reproduced. Press a "pause" control and sing or play and the room is as it was, in its natural state. However, the prime objective here is to maximize the quality of reproduced sound and in that it is possible to succeed using multiple subwoofer methods. Bass traps are optional.

These are considerations that need to be incorporated into a home theater plan at an early stage so that limitations imposed by real-world circumstances can be factored into the choice of a solution.

4 Level and Delay Adjustments

There are a few adjustments that need to be made to a system before the "play" button is pressed. All of them apply only to the prime listening location—the "money" seat in installer parlance.

- The sound level at the prime location should be set to the reference level (usually 75 dBC for domestic equipment) and it should be the same for all channels. There are many ways this could be done using external apparatus, but receivers and surround processors incorporate test signals for this purpose, along with instructions on setting reference levels, sometimes with a provided microphone. Significant research has been done on how best to do this (Section 14.2.2, p. 388) and the result is encouraging in that there is considerable agreement on techniques that work—the common band-limited pink noise signal is one of them. Obviously, the L, C and R channels need the closest attention. It is unlikely that any of the common procedures would result in errors sufficient to be audible in program, much less to degrade entertainment.
- Sounds from all channels should be adjusted to arrive at the prime location at the same time. This is done using delay capabilities built into receivers and processors. This is most critical for the L, C and R loudspeakers, where fractional millisecond (foot or meter equivalents) need to be attended to.

None of this is complicated, but it must be done if the system is to operate as intended. But, bear in mind that all of these "precision" adjustments only work for the prime listener—everyone else in the room hears imperfectly calibrated sound. Fortunately, for the most part, it is not noticed.

Playback at reference level for movies can generate 105 dB SPL from each loudspeaker. Combining multiple channels the overall sound levels are well within the range of potential hearing loss. It also may well exceed the power output capabilities of the power amplifiers and the power handling capabilities of common loudspeakers.

Experienced listeners have found that in home theaters as well as cinemas playback at reference level can be unpleasantly loud. Some cinema owners, responding to customer complaints, turn the volume down by as much as 10 dB, which relieves one problem, but can result in reduced dialog intelligibility, thus creating another. This is evidence of what has come to be called the "loudness wars," made possible by the additional undistorted dynamic range that digital sound tracks offer over the old analog systems. The professional cinema industry is looking at ways to improve the situation, but it is not a technical problem; it has to do with judgment and taste. Personally, I think that tiresome loud rumbles and sound effects are sometimes used to offset a limp plot, but I am not the target audience.

In home theaters, playback at less than reference sound level is very common; however the consequences are different because in the repurposing of sound tracks for distribution outside the cinema world (discs, TV and streaming) dynamic range is frequently reduced and dialog levels may also be adjusted to be more intelligible. This is to accommodate the masses of small, low power systems in homes. However, if one is determined to play at reference level, significant power in power amplifiers, and power handling in loudspeakers will be necessary to avoid distortion, current or voltage limiting, and finally protection circuit activation. See Part 3 in this series.

Home playback also permits subtitles, which many people, even young viewers, find helpful in some films. In the end, ignore the volume number being displayed and play movies at levels that do justice to the dramatic sound effects but that do not become an annoyance. This is obviously highly personal and it will be program dependent.

NOTE: Well-chosen smartphone apps can perform adequately accurate measurements for sound level calibrations. See: B. M. Faber, "Acoustical Measurements with Smartphones: Possibilities and Limitations," *Acoustics Today*, Summer 2017, vol. 13.

5 Equalization—What Works and Does Not Work

Many consumer audio receivers and surround processors incorporate some form of "room equalization." Currently there are a few competing branded algorithms and some DIY options. They operate in different ways, but all of them assert the ability to improve sound from any loudspeaker in any room. This is an overly ambitious claim but one with proven market appeal. In my teaching at CEDIA events I asked several classes of installers what their experience has been. Remarkably few were supportive, most had reservations, and some claimed that the best setting is "off." Internet audio forums have animated debates about the relative virtues of different proprietary algorithms.

This is as it must be. No two combinations of rooms and loudspeakers are the same, and different algorithms are more or less successful at doing the good things and not doing the bad things.

Let there be no doubt about it: equalization can be very useful. But it is not *universally* useful. It is an *essential* part of delivering satisfactory bass in small resonant rooms, as discussed in Chapter 8. But, above the transition frequency, 300–500 Hz or so, the situation changes. In powered loudspeakers or loudspeakers with dedicated electronics, using equalization based on anechoic measurements, good loudspeakers can be made better. But, if all that are available are in-room measurements, things may be seen that are alarming to the eyes but that may not be audible to the ears. “Correcting” them may degrade a good loudspeaker. Other features seen in room curves may truly be problems, but without comprehensive anechoic data on the loudspeakers it is impossible to know what caused the problem and whether equalization is the appropriate remedy. Sometimes the only solution to getting good sound is to acquire some properly designed loudspeakers.

Figure 4 illustrates the problem: two ears and a brain respond to the complex sound field in a room in very different ways than an omnidirectional microphone and analyzer. In the common steady-state measurements, the omnidirectional microphone does not discriminate in either direction or time of arrival of sounds. Two ears and a brain are highly responsive to such differences, creating perceptions that have identifiable sound quality, directional and spaciousness components. One can do time-domain measurements—impulse or energy-time curves (ETC)—but again, incident angle is ignored. The measurements do not include all of the factors that matter to human listeners, and therefore they cannot be reliable predictors of perceptions.

A single curve cannot describe a loudspeaker in anechoic measurements; neither can a single room curve describe the multidimensional sounds within a room. It should be common sense, but it seems not to be.

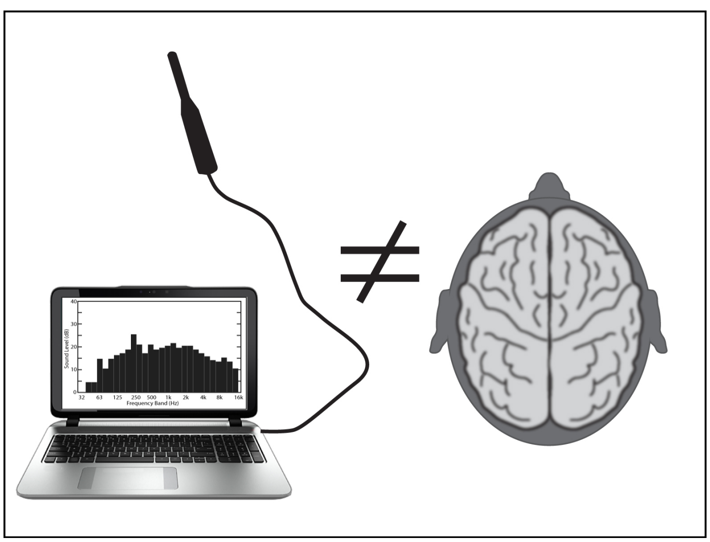


Figure 4. A simplistic expression of the fact that two ears and a brain perceive much more than an in-room steady-state measurement with a microphone and analyzer can possibly indicate. They are not equivalent evaluators of sound or spatial qualities.

The problem with many room EQ algorithms is that they respond to non-minimum-phase irregularities (acoustical interference/comb filtering) in the room curves by smoothing them. In the process they risk degrading the performance of truly excellent loudspeakers. If a rough “before” curve is transformed by auto-EQ into a smooth curve (at the same frequency resolution) one can reasonably speculate that something improper has been done.

The fact that some algorithms offer several optional “target” curves is an open admission that the room curve is not a definitive statement of sound quality. When they offer the capability of changing the shape of the target curve to improve sound quality, it is an admission that this is a subjective, trial-and-error multi-filter tone control adjustment to make the program(s) of the moment sound good, not a technical calibration process. Because it is unlikely that adequate anechoic data are available on the loudspeakers being used, nobody can know what is being compensated for: the loudspeaker frequency response, the loudspeaker directivity, something acoustically aberrant in the room, or the program being auditioned during the trial-and-error subjective equalization process.

Even the most neutral reproduction systems can produce unsatisfactory sounds. They are at the mercy of the program material; they just “tell it as it is” but without unnecessary added colorations. Because of variations in program, caused in part by the circle of confusion (Section 1.4, p. 9), tone compensation is a necessary part of sound reproduction for those listeners who hear and are bothered by too much or not enough bass, treble, etc. Such variations can sometimes be heard within the same album, as decisions were made during the recording process to appeal to the bass-deprived masses, delivering too much bass to customers with well-balanced systems, or to compensate for excessive bass in the control room monitors, leaving the customer with insufficient bass. For those of us who hear these things, tone controls need to be available at all times. Sadly, audiophile purists, in their ignorance, years ago assumed that recordings are perfect and that tone controls should be banished from high-end products. Fortunately, I note that some high-end preamplifiers have reinstated tone and tilt controls. All should. Some products that incorporate tone controls force the user through a menu to access the controls and then impose program interruptions while the digital system implements the adjustments. It is not a perfect world. I want an analog-equivalent set of controls available on my remote control.

These room equalization topics are discussed in detail in Sections 5.8, p. 142, 12.2.3, p. 348, and 13.2.3, p. 371. The short summary is that if one begins with well-designed loudspeakers there Is little or nothing to be done above a few hundred Hz. Unfortunately, the public has been led to believe that full-bandwidth equalization is essential, and that these algorithms can transform any loudspeaker in any room into something perfect. *Caveat emptor.*

6 Baffle Walls

Quality cinemas often build the front loudspeakers into a “baffle wall,” a smooth surface that is curved just enough that the flush mounted loudspeakers are properly aimed at the audience. Figure 5 illustrates what might be the definitive baffle wall. Needless to say, it was well constructed—not all are.

Figure 5. The Samuel Goldwyn Theater in the Academy of Motion Picture Arts and Sciences, Hollywood, California with the screen down and showing the five front channels and subwoofers, all JBL loudspeakers. Photo courtesy of JBL Professional; "Academy Award" and "Oscar" image © AMPAS ®. THX® Lucasfilm, LTD.

The baffle wall provides a 2π half-space mounting for the woofers and a barrier preventing sound reflected backwards from the perforated screen from rattling around in the sometimes large, reflective space behind the screen. Some baffle walls do not cover enough of the opening and allow audible reflected sounds to leak from the reverberant space behind. Older cinemas, converted from music or drama halls, can have large, high spaces behind the screens. Obviously, the baffle wall must be covered with at least two inches of fibrous material or acoustical foam to absorb the sound reflected from perforated screen.

All of this is interesting, but is a baffle wall necessary in home theaters, however much it adds "authenticity" to the installation? First, it is unlikely that there is a large reflective space behind the screen. However, if for any reason the screen is located far away from the front wall, a baffle wall is necessary. Otherwise there is a choice. Sound reflected from a perforated screen can be absorbed using fibrous or foam absorbers on the front wall. The front wall is a good location for absorbing materials in any event (See Figure 2, in Part 1). If a woven screen is used, less sound is reflected backwards. The loudspeakers are likely to be very close to the front room wall, meaning that at the long wavelengths involved, 2π solid angle gain at low frequencies is present. The only remaining issue is the adjacent boundary acoustical interference, as illustrated and explained in Section 9.3, p. 269. For this there is no perfect remedy. There will be a narrow non-minimum-phase dip in the direct sound, which is risky to equalize, but the steady-state effects are much less severe. These effects usually fall into the frequency range of other adjacent boundary and early-reflection irregularities and may or may not be visible or audible as an identifiable effect.

In perusing some Internet photos of baffle-wall installations I have seen examples of bad practice in that the loudspeakers were placed inside enclosures sunk into the wall, and the cavity around the loudspeakers were not closed off or filled. Figure 9.9, p. 272 shows what might happen. The loudspeakers should be flush mounted without gaps. This achieves a perfect 2π mounting, which is beneficial to the bass output, but then there is the matter of the changed diffraction around the edges of the enclosure at higher frequencies. The edge diffraction of enclosures is part of the loudspeaker design and removing it or changing it actually changes the loudspeaker design. This is more likely to be a problem with cone/dome designs than designs using large horns, but it cannot be ignored.

I would be inclined to avoid a baffle wall for loudspeakers designed to be free/floorstanding because of the diffraction issue, which affects off-axis performance. Equalization can compensate for the bass rise, as it would have to in the baffle wall case, and also the adjacent-boundary dip due to wall behind. Baffle walls are a "cinema" thing, not a requirement for routine domestic installations. In any case, not less than 2" (50 mm) of absorbing material on the front wall behind a perforated screen is essential.

Loudspeakers designed for in-wall, 2π mounting have no such issues, and in smaller rooms the best of these loudspeakers can perform admirably in all locations. Other than being compromised by a rigid attachment to what is likely to be a wall designed for structural not acoustical performance, there is no reason why in-wall loudspeakers should radiate inferior sound.

Last modified October 16, 2017

Designing Home Theaters and Listening Rooms:
Part 3—Power Amplifiers—How Much Power is Needed?

by

Floyd E. Toole, PhD

Outline

The 500-page book *Sound Reproduction—the Acoustics and Psychoacoustics of Loudspeakers and Rooms*, by Floyd E. Toole, Focal Press, 2017 has in-depth discussion and data on many topics. In this much-simplified guide readers will be provided with some additional information, and directed to parts of the book using figure and section numbers and page numbers for more explanation.

Other documents on this website will also be referred to. The website will change with time, so updates to this guide will follow; check the edition date at the end.

1 Introduction

To determine how much amplifier power is needed in a home theater aspiring to be able to reproduce calibrated cinema sound levels is simple in theory, but complicated in practice. The reason is that the information we need is often not available or, if available, it is incomplete or not trustworthy. In the following sections I will show some examples of problems related to amplifier power ratings, and loudspeaker impedance and sensitivity ratings.

For music reproduction there are no standardized sound levels, so the determining factor is revealed only when the customer can or cannot play the music of choice at the level of choice without gross distortion or expensive silences.

Some amplifiers are able to drive almost any loudspeaker, others can be temperamental—it would be good to be able to anticipate the tolerant ones in advance. We are not talking about the subtle differences. When a power amplifier goes into protection mode, the results are plainly audible. At levels approaching protection, increased distortion may be evident. It is not just a matter of having inadequate "rated power." There are other factors involved, and unfortunately specification sheets do not provide the answers.

There are several significantly different power amplifier design configurations. Tube/valve amplifiers, class A and class A/B solid state and class D “digital” amplifiers all have admirers and detractors. Used within their limits, no well-designed power amplifier is likely to interfere with the enjoyment of music and movies. That said, there are technical factors, such as power output, current capability, size, weight, heat, fan noise, etc. that may favor one design over another in specific applications.

Less obvious is the internal impedance of the power amplifier, which is not normally specified directly, but instead as damping factor which is 8 (ohms) divided by the internal impedance. As explained in Chapter 16, p. 433, the damping factor has nothing really to do with damping loudspeakers, but it does have a lot to do with how much the frequency response of the loudspeaker is changed by the power amplifier.

Solid-state amplifiers typically have output impedances in the range 0.01 to 0.04 ohms (damping factors from 800 to 200). Tube amplifiers typically range from 0.7 to 3.3 ohms (damping factors from 11 to 2.4), and occasionally even more. These are large losses, and when placed in series with the frequency-dependent impedances of loudspeakers, they cause audible changes in spectrum; the loudspeakers have essentially been “revoiced” by the tube power amplifier. See Figures 16.1 and 16.2, pp. 434 and 436. *Stereophile* magazine measurements of power amplifiers include frequency responses when driving a simulated loudspeaker load. The deviations are sometimes substantial. Somehow this is not discussed in the subjective reviews, when what they are hearing is through loudspeakers that have been modified in ways they do not know. They are not the products the manufacturers designed.

As a general statement, it is advisable to leave tube/valve amplifiers to those who believe that their sound is as satisfyingly warm as the air above them.

The fact is that *well-designed* solid-state amplifiers have no fundamental problems. Loudspeakers will sound as they should, and power and current capabilities are available to satisfy all needs. Audible differences between good power amplifiers used within their limits are vanishingly small, usually inaudible. Amplifier sound quality has been one of the “great debate” issues for years. Double-blind listening tests conducted over several decades have been very disappointing to those who thought there were easily heard differences. A common reaction has been that the problem was the blind test, even though there was no time limit, and a free choice of program. Some said that if they knew what they were listening to, it would have been different . . . of course. That said, there is no denying that inadequate engineering is sometimes evident, which is why the final test of an amplifier is a subjective one, with it driving the customer’s loudspeakers. For the technically inclined, watching an oscilloscope display of the voltage across the loudspeaker terminals is sometimes greatly revealing, especially when power amplifiers approach voltage or current limits and go into “protection” mode.

Loudspeakers are complex electrical loads, exhibiting magnitude and phase variations as a function of frequency. No two are alike. Power amplifiers are most often rated and tested using purely resistive loads. How an amplifier behaves when connected to a real loudspeaker load is difficult to predict, especially from published specifications. Very high current delivery capability is a useful indicator, as is a stated ability to drive very low impedances, like 2 ohms. At moderate listening levels, there are usually no problems, but when pushed close to the limit there may be misbehavior. Trial and error is a necessary part of this kind of experience, and this, as much as anything, explains the attraction to monster power amplifiers with large current and voltage capabilities—a.k.a. safety margins—a.k.a. headroom—a.k.a. peace of mind—at a price.

In the mass-market audio world budgets are finite and value for money is an important factor. The basis of most home theaters is a receiver, and it is remarkable how little the *advertised* power amplifier output has changed even though the channel count has gone up, and the size and weight of the units seems not to have changed much. Some of that is attributable to new amplifier and power supply technologies, but there is more to the story.

Chapter 16 from p. 433 discusses some important amplifier/loudspeaker compatibility issues. It starts by pointing out that loudspeaker impedance varies with frequency, and as a result the frequency response of loudspeakers is altered when there is significant resistance in the signal path. The obvious source of resistance is the speaker wire, where the cure is an inexpensive one of choosing a large cross-section wire (i.e. a low gauge number; 12 being sufficient for most domestic runs). In spite of advertising claims, assisted by some susceptible (or plainly cynical) audio journalists, exotic speaker cables sound no better than wire from the local hardware store. If it is to be pulled through conduit, there are cables with coverings that make the job easier and which may be more resistant to the high heat of attics, but this has nothing to do with sound quality.

2 Power Ratings of Amplifiers

These should be simple to understand, but they are not. The main reason is that the power rating depends on several factors such as:

- the complex impedance (magnitude and phase angle) of the load that the amplifier is driving,
- the frequency range/bandwidth of the signal being used, ranging from a single frequency to full bandwidth.
- the duration of the signal, ranging from continuous to “peak/instantaneous.”
- the distortion level decided upon to represent that a limit has been reached.

These variables are easily manipulated to generate numbers for specification sheets that are almost always optimistic. One is often left with some basic questions after reading specification sheets.

I just looked up the specifications on a few popular 9-channel immersive audio receivers. The only output specification that addressed the full frequency range, 20 Hz to 20 kHz, was for two channels driven: stereo. The maximum output rating was for a *single* channel driven at a *single* frequency of 1 kHz—this is a useless rating employed only to be able to show a large number on the spec sheet. Both of these were into 8-ohm resistors. Real loudspeakers are not resistors, and many of them exhibit impedances that drop to 4 ohms or less. I found no measurements with more than two channels driven and no measurements of any kind made with impedances lower than 6 ohms. There were no warnings about minimum impedances (an indication of maximum current delivery). Many (all?) of these devices work from a single power supply and the more channels that are simultaneously operating the less current that is available to any one of them.

It is likely that these amplifiers will experience distortion, voltage or current limiting, or protective shutdown, when attempting to play multichannel programs at anything resembling cinema reference level. (It is stated in some of the owner's manuals that the built-in calibration is referenced to cinema level.) It is fortunate that most people don't listen at close to cinema sound levels at home.

But there are people who do want to play at or close to reference level—including me—to experience the full dynamics of blockbuster movies. What then? This is what line-level surround processors and stand-alone power amplifiers are for.

By tradition, amplifiers are rated according to how much steady-state power they can drive into an 8-ohm pure resistance. This is sometimes labeled "RMS" indicating that it was calculated using the rms value of voltage. If this is achieved with low non-linear distortion levels (usually THD) the indications are good. But, the real-world loads for amplifiers are loudspeakers, which are not pure resistors, exhibiting both inductive and capacitive reactance. In addition, the impedance is not constant at all frequencies; a loudspeaker system rated at a nominal 8 ohms might vary from a low of 3 ohms to a high of 20 ohms or more at different frequencies—the higher numbers are not problems, but the lower ones may be. And, of course, music and movies are not monotonously at full output for lengthy periods, so this rating is very conservative, yielding a safe number.

Recognizing this, some manufacturers display a "peak power" rating, where more output may be available for a short "transient" interval, after which it reverts to the steady-state level. Interestingly, the amount of this peak power "headroom" depends on the design of the power supply. In amplifiers with "stiff," voltage-regulated power supplies the peak power rating is double the continuous power rating—a headroom of 3 dB, because a pure tone has a peak to average power rating of 3 dB.

However, in amplifiers with "loose," unregulated power supplies, large filter capacitor banks can supply current for short transient demands at levels much higher than for a steady-state demand that draws the power supply voltage down, resulting in a reduced continuous power output. Such amplifiers can legitimately show large peak power capabilities, but these additional momentary watts may or may not be useable. As will be seen, movie sound tracks are highly standardized. The maximum sound level is represented by a pure tone generating a potential sound level of 105 dB at the reference listening position. This corresponds to a peak of 108 dB SPL, 3 dB higher. It also corresponds to 0 dB digital program level, above which there is no signal. So, if an amplifier meets the 105 dB continuous pure tone requirement, the +3 dB peak level is automatically achieved, and there is no advantage to having more "headroom"—it won't be used for movies.

However, for music there are no rules. There is no right or wrong here, just differences that might matter. Peak power capability can be measured by integrating the output over a specific short time interval. In the absences of accepted standards, different manufacturers have chosen different intervals, ranging from 10 ms to 500 ms in my limited explorations. These are probably attempts at being usefully revealing, but, again, the resulting numbers cannot be compared to each other or necessarily to real-world demands.

Finally, there are the totally ambiguous "music power," "program power" claims, or just a number, with no qualifications. These are totally without definition, and in the past were used by unscrupulous marketing departments to inflate power ratings. At a point, the FTC (the US Federal Trade Commission) stepped in and attempted to influence manufacturers to employ a standardized method. It was not perfect (no single number is) but it did gain a following, and seemed to tame the most outrageous claims. So the FTC rating is yet another number to be seen in spec sheets. It is a specific kind of continuous power rating embracing the full bandwidth. IEC (International Electrotechnical Commission) ratings are also seen. These are tested only at a single frequency.

In conclusion, there is no universal standard that is used for specifying power output. Given these observations, a prudent person would choose power amplifiers using a very "safe" interpretation of the specifications: in other words, allow some "headroom."

3 Sensitivity Ratings of Loudspeakers

This, unfortunately, is another area in which getting the right answer is difficult. Manufacturers understandably do what they can to make their products look good. Here the tendency is to imply that the loudspeaker impedance is easy for power amplifiers to drive and the sensitivity is high enough that a lot of drive may not be necessary.

However, not all discrepancies are the result of marketing exaggeration. There are legitimate problems that show up in sensitivity specifications and in reviews where sensitivity is measured. Figure 1 shows a simplified example. The measurement should be made in an anechoic chamber or a reflection free FFT time-windowed functional equivalent. The dotted curve is the on-axis frequency response of a mediocre loudspeaker with substantial variations in level at different frequencies. A manufacturer might pick the highest point in the curve because it yields the highest number, but even trying to be honest is a problem if the frequency response is not smooth and flat. In the example, I show the B-weighting curve used by John Atkinson (*Stereophile* magazine) because at one time it was thought to be a good measure of overall loudness. That has now changed—see Section 14.2, p. 385. The 300 to 3kHz range is used in the NRCC tests I originated (www.soundstage.com), and the 500 to 5 kHz range is used in *Sound and Vision* tests. The latter two avoid the frequency range where room resonances and adjacent-boundary effects are factors in the real world. I think these are more realistic. However, the real point is that when the frequency response is significantly non-flat, the frequency range over which one averages the sound output can bias the sensitivity rating up or down. Most manufacturers do not specify *how* they measure sensitivity, so it is no surprise to find disagreements between published sensitivity numbers and those resulting from independent tests.

Obviously, the smoother and flatter the on-axis frequency response, the greater will be the agreement among the different methods of measuring loudspeaker sensitivity.

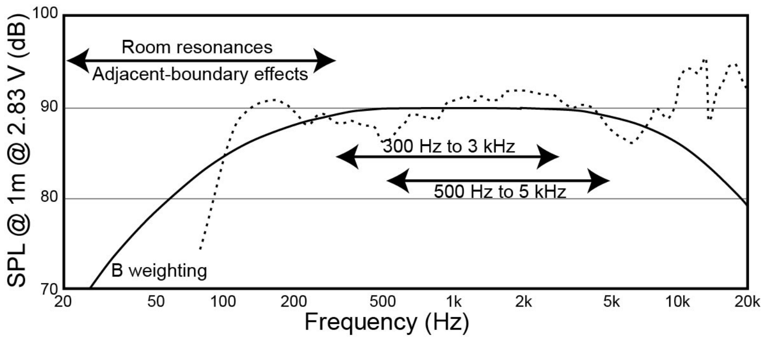


Figure 1. An illustration of how variations in sensitivity ratings can result from measuring over different portions of the frequency range. The input signal is a standardized 2.83 V.

To provide an example, I surveyed 55 loudspeakers measured by soundstage.com, who use the measurement facility at the NRCC, my old lab in Ottawa, Canada. I created the original measurement system, and trust the data. I was interested in two simple things, the magnitude of the impedance and the sensitivity. Both of these are essential information if we are to attempt to predict power amplifier needs. I then looked up the manufacturers' published data, which is what consumers and consultants would use for guidance.

Some of the comparisons were quite close, within a dB or so, and not surprisingly they were for loudspeakers having relatively smooth and flat on-axis frequency responses. Non-flat loudspeakers showed greater variations, as expected, but then there were variations with no obvious explanation other than simple deception; a few were quite large. Figure 2 shows a distribution of the deviations found in the survey, along with the multipliers for amplifier power associated with the deviations. It can be seen that a factor of two was common (twice as much amplifier power is needed than might have been anticipated from the published sensitivity). One product was in error by a factor of 5. Presumably the marketing department selected the sensitivity.

Adding confusion is the frustrating use of the obsolete SPL @ 1 m @ 1 W, when one suspects that it is the voltage sensitivity that is measured, as it should be (see Section 16.3, p. 437).

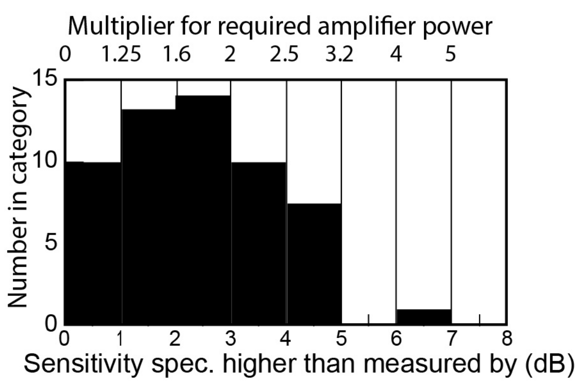


Figure 2. The difference between published sensitivity specifications and sensitivity measured over the 300 to 3 kHz frequency range as done at the NRCC (extracted from data on www.soundstage.com*). The numbers at the top show the amount by which the published specification underestimates the size of power amplifier needed.*

These are not trivial discrepancies, and different ones, some higher, some lower, will be found if one looks at comparisons of published sensitivities compared to those in reviews by Stereophile, *Sound and Vision*, Audioholics.com and others. It is simply an unfortunate fact that makes our lives unnecessarily complicated. As we move forward to an amplifier power prediction, it is obvious that there already is a significant problem.

4 Impedance Ratings of Loudspeakers

The voltage sensitivity ratings of loudspeakers assume an 8-ohm load. Many loudspeakers are specified as having 8-ohm nominal impedances, but as illustrated in Figures 16.1 and 16.3 (pp. 433–436) reality can be very different. The interaction between power amplifiers and loudspeakers is complex—beyond my competence to discuss. In addition to the magnitude variations in impedance, there are phase variations, the combination of which can cause some amplifiers to go into protective limiting in some combinations of signal and loudspeaker load. The topic is well examined in: Benjamin, E., "Audio Power Amplifiers for Loudspeaker Loads," *J. Audio Eng. Soc.*, vol. 42, pp. 670–683, 1994. There is a long list of references for more elaboration.

Although specification sheets most often cite only a single, nominal impedance number, it is a definitive criterion only if the impedance does not vary. A few—very few—loudspeakers electrically compensate for variations in impedance. It can be done, but it costs money, sacrifices some sensitivity and may compromise frequency response—for a feature that average consumers do not appreciate as an advantage.

Fortunately, the minimum impedance magnitude is a useful indicator of potential problems and knowing this, some manufacturers state a minimum impedance as well as a nominal impedance. In the soundstage.com survey I did:

- The average *specified* nominal impedance for all loudspeakers was 5.9 ohms, ranging from 4 to 8 ohms.
- The average *measured* minimum impedance for those products was 4.6 ohms, ranging from 2 to 10 ohms.
- The average *specified* minimum impedance stated by those manufacturers that declared it was 4.54 ohms, ranging from 2.7 ohms to 8 ohms.
- The average *measured* minimum impedance for those products was 4.4 ohms, ranging from 2 ohms to 10 ohms.

The conclusion seems to be that selecting power amplifiers capable of driving 4 ohms is a good idea and, for some products, a 2-ohm capability is a necessity. These requirements are factors of two (4 ohms) and four (2 ohms) in power output compared to the 8-ohm reference load.

To illustrate just how unrealistic the present situation can be, Figure 3 shows an expensive, respected, high-end loudspeaker, rated by the manufacturer at 8 ohms. It is 8 ohms at only four frequencies, and elsewhere it is much lower. As a result, at the standardized 2.83-volt input, the loudspeaker is drawing much more current, and therefore is delivering much more power, than a true 8-ohm load would demand. It is a misleading specification. Many power amplifiers including almost all those in receivers cannot drive 3-ohm loads without risk of activating their protection circuits at high sound levels. Many receivers would have problems with 4-ohm loads, but this is not always stated in the specifications. This is a regrettable lack of candor on the part of this loudspeaker manufacturer and of the receiver purveyors. Sadly, it is common.

Loudspeaker sensitivity used to be specified as the SPL at a distance of 1 m with an input of 1 watt. Power is voltage squared divided by resistance (Ohm's law) so because loudspeaker impedance varies with frequency that kind of rating is not practical. It was eventually changed to a voltage sensitivity: SPL @ 1 m @ 2.83 volts. 2.83 volts delivers 1 watt to an 8-ohm load.

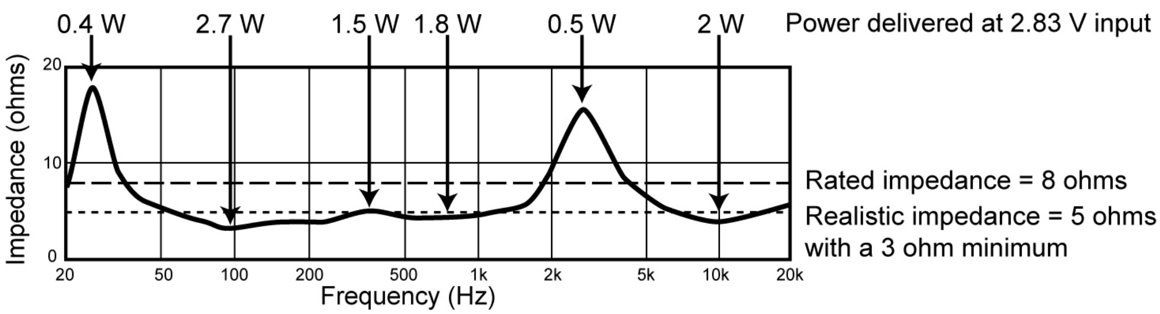


Figure 3. A typical loudspeaker impedance curve, showing the rated impedance of the product and, at several frequencies, the input power for 2.83 V. Figure 16.3, p. 438 in the book.

Modern solid-state power amplifiers are well described as "constant-voltage sources." They attempt to maintain a specific voltage across the output terminals even when the load impedance varies, as it does with frequency, and when using loudspeakers having different nominal impedance ratings (8, 6 and 4 ohms are common). Thinking simplistically, if one is listening to an 8-ohm loudspeaker and exchanges it for a 4-ohm loudspeaker without changing the volume setting (the same voltage signal is appearing at the output terminals of the power amplifier), then the 4-ohm loudspeaker tries to draw twice the current from the amplifier, and if successful, will dissipate twice the power (power = voltage × current). Suddenly a 100-watt (rated at 8 ohms) amplifier driven to its limits is required to deliver 200 watts into a 4-ohm load. The question is: can it succeed? The answer is that it depends on how the amplifier has been designed. Most are able to drive the 4-ohm load but at less than double the power output.

My first "super-amp," as they were called many years ago, had a hair-trigger protective circuit, occasionally inserting ugly sounds into loud passages, so I disabled it. I had to replace a few unprotected output transistors before moving on to a better design. Another amplifier did the same thing with certain loudspeakers, but the manufacturer was much more receptive to my complaint. They modified the circuit and offered free upgrades to customers of the recently released product. I wonder if this insidious problem still exists. One just hears unpleasant additions to the sound at high levels, and identifying the cause is difficult. Sometimes loudspeakers take the blame. There are no specifications for such things. Different amplifier designers choose different solutions, from "beefing up" the amplifier so that it does not need much protection, to designing ever more elaborate protection circuits that avoid false triggering. But, the reality is that before the circuit triggers to protect the amplifier, it is probable that distortion is occurring.

Another set of design variables determines how an amplifier sounds at clipping because the beneficial effects of feedback are substantially lost. Clean clipping up to about 6 dB is difficult to hear (Section 16.3, p. 438), but many designs misbehave when they clip, generating spurious signals during even moderate clipping that have detrimental audible effects.

A little audio history: many years ago, a reviewer noted that a loudspeaker he was reviewing "revealed" difference in power amplifiers. It was a sensation. Upon further investigation, it was found that the loudspeaker was incompetently designed, with an impedance that dropped to under 1 ohm. The loudspeaker did not *reveal* differences between power amplifiers, it *caused* the differences. The amplifiers in question were just fine. But, no doubt it did wonders for sales of monster "arc-welder" monoblocks.

So, as we progress to an estimate of required amplifier power we have accumulated another uncertainty. Because the reactance of loudspeakers is not normally specified, and the tolerance to load reactance in power amplifiers is not specified, it is wise to allow some headroom.

5 Calibrated Sound Levels in Cinemas and Home Theaters

There are no "calibrated" sound level references for recording or reproducing music. We set the volume control where we like. With movies, though, there is a "reference" playback level that is used in dubbing stages and postproduction studios where soundtracks are created, and in cinemas where they are reproduced. This reference level is maintained in home theaters. Whether one chooses to play movies with the volume control at "0" is an option. Many people find it is uncomfortably loud with some movies.

As a result, home theater playback at less than 0 dB volume-control setting is commonplace. Even a small reduction like 3 dB, a small change in subjective loudness, is a factor of two reduction of sound power output. Instead of 200 watts, one is using 100 watts. 10 dB is a factor 2 in perceived loudness and a factor of 10 in power: from 200 watts to 20 watts. Perhaps this explains how so many people appear to find happiness with modest receivers.

The professional systems are calibrated at a steady-state level of 20 dBfs (fs = full scale), which corresponds to 85 dB, measured using broadband pink noise and a C-weighted sound level meter at a seat 2/3 of the distance between the screen and the back of the cinema. This is with a playback system (the B chain) calibrated to the X-curve, which has a steady-state target response that is flat from about 50 Hz to 2 kHz and then rolls off at 3 dB/octave. It needs to be noted that home theaters should not be calibrated for the X curve. Soundtracks are repurposed for disc, streaming and television distribution and are supposed to be suitable for playback through "high fidelity," flattish direct sound audio systems.

Sound tracks were given 20 dB of headroom for dramatic effects, giving us the potential of 85 + 20 = 105 dB SPL maximum sound levels at the reference listening position. This is illustrated in Figure 4.

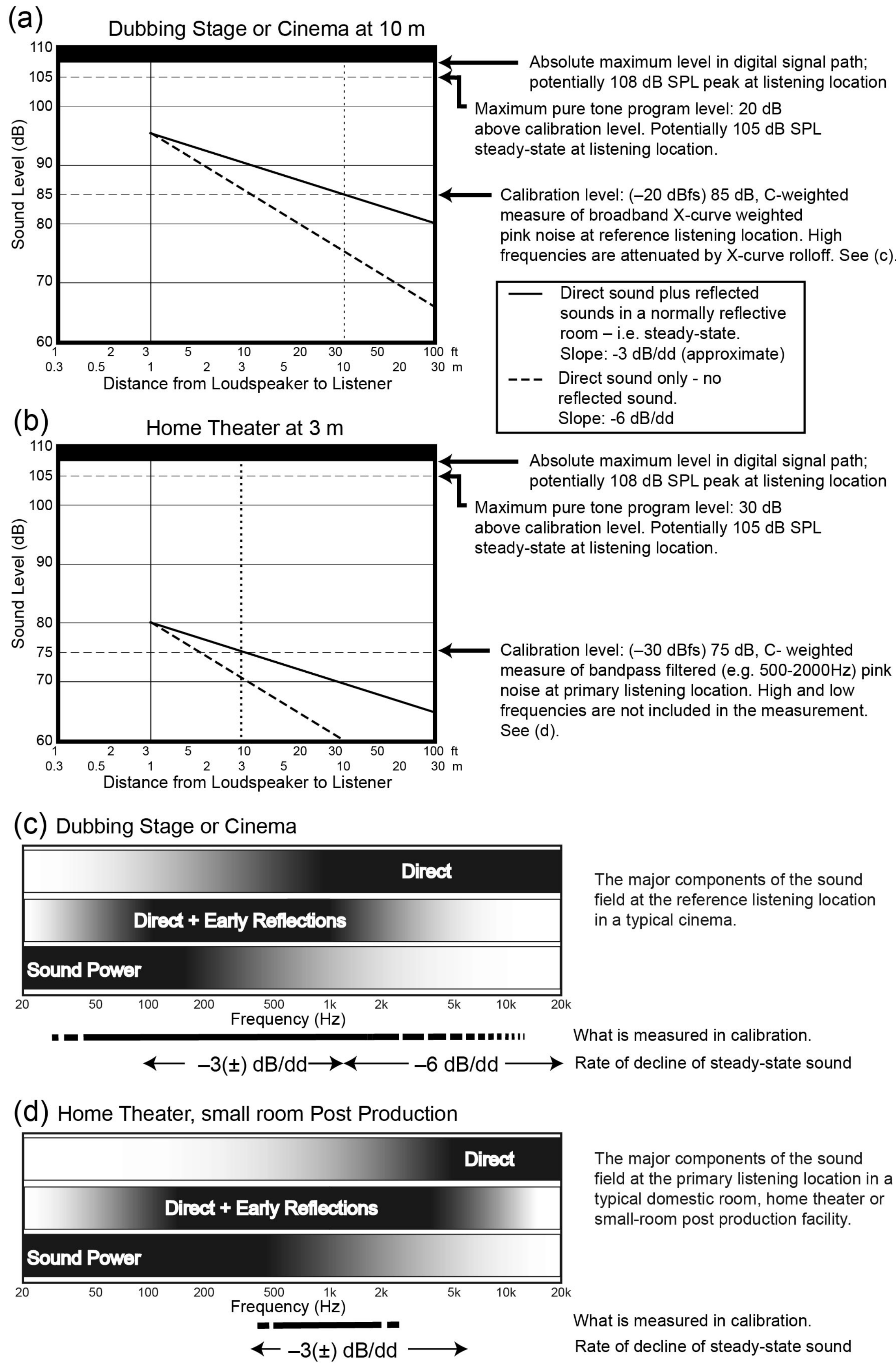


Figure 4. A graphic illustration of the sound levels in cinemas and home theaters, including explanations of the signals used for the calibrations.

The current digital cinema package (DCP), which includes audio, video and other data, is very different from the old analog systems. Here a digital audio signal has a finite upper limit (0 dBfs) and up to that limit signal quality is uniformly high. As a result, soundtracks tend to be recorded at higher overall levels than in analog days, but the absolute maximum level is still a momentary peak that cannot be higher than 0 dBfs in the line-level signal path. If the maximum sound level defined by a pure tone is 105 dB SPL, the peak level is 3 dB higher, which corresponds to 108 dB in a calibrated venue.

If each channel is capable of generating 105 dB SPL, obviously the combination of L, C and R and the surround and immersive channels can produce much higher sound levels. With digital soundtracks and the aggressive sound tracks in some blockbuster films, these can reach rock concert sound levels. This has motivated enough moviegoers to complain to cinema managers that they reduce the playback levels—by as much as 10 dB for some movies, which usually creates a problem with dialog intelligibility. Establishing some control and order in this situation is a current topic of discussion in SMPTE and AES standards groups. It will be difficult.

In home theaters the calibration sound level has been reduced to 30 dBfs, which should deliver 75 dB, measured with C weighting at the prime listening location using a band-limited pink noise signal that is usually built into the receiver or surround processor. After calibration, with the volume control set to 0 dB, the maximum sound level is theoretically 105 dB (108 dB SPL peak) for each channel, including surround and immersive channels. In practice the most aggressive sounds emerge from the screen channels and the subwoofers (which include the LFE).

6 Estimating Amplifier Power Requirements for Home Theaters

The following is a simplification of important realities all of which cannot be known. However, these are the data given to us by manufacturers, and although they are incomplete, or perhaps not even accurate, they can provide guidance. Based on observations, some of which are discussed in the preceding sections, errors are likely to result in underestimating amplifier power requirements. Consequently, whatever number is yielded by the following procedure, it would be prudent to consider it as a minimum requirement and to allow for some headroom.

First, we need to estimate the sound level the loudspeaker must radiate at the reference distance of 1 m (where the sensitivity is specified) to deliver the calibration sound level to the prime listening position in the home theater.

We know that loudspeakers are small compared to the distances at which we listen to them, so it is reasonable to think that they behave as “point sources” radiating an approximately spherical wave front, the sound level of which decreases at a rate of about 6 dB for each doubling of distance (−6 dB/dd). This is the well-known “inverse-square law” illustrated in Figure 5, where it is seen that close to the loudspeaker the sound level falls away very rapidly, and, as distance increases, the decline is less rapid.

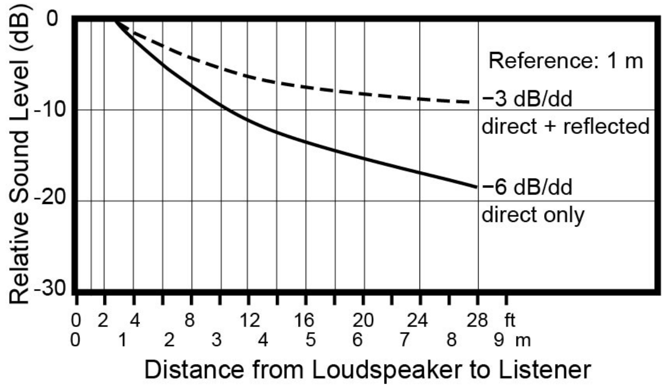


Figure 5. The sound levels as a function of distance from a typical loudspeaker in a typical room. Shown are the curves for the direct sound alone, declining at a rate of −6 dB/double-distance, and for the direct and reflected sounds combined, declining at a rate of −3 dB/dd. Figure 14.3, p. 393.

This is the direct sound, the first sound to arrive at a listener's ears. It is perceptually the dominant factor for sound localization and very important for sound quality as well. However, reflected sounds contribute most of the sound energy that we measure in small home theater rooms. Therefore, we also need to examine how the sound field, including reflections, changes with distance from the loudspeaker. In Figure 5, this is shown as the −3 dB/dd (double distance) curve. In reality, the slope varies from about –2.5 dB/dd to −3.5 dB/dd (see Figure 10.8, p. 294).

In terms of perceived loudness of sounds and calibrated sound levels the –3 dB/dd curve is the one that matters.

In acoustically "dead" rooms something closer to the –6 dB/dd curve may be more representative, but such rooms are definitely not advised for recreational listening. Overly damped rooms do not sound pleasant, even to converse in.

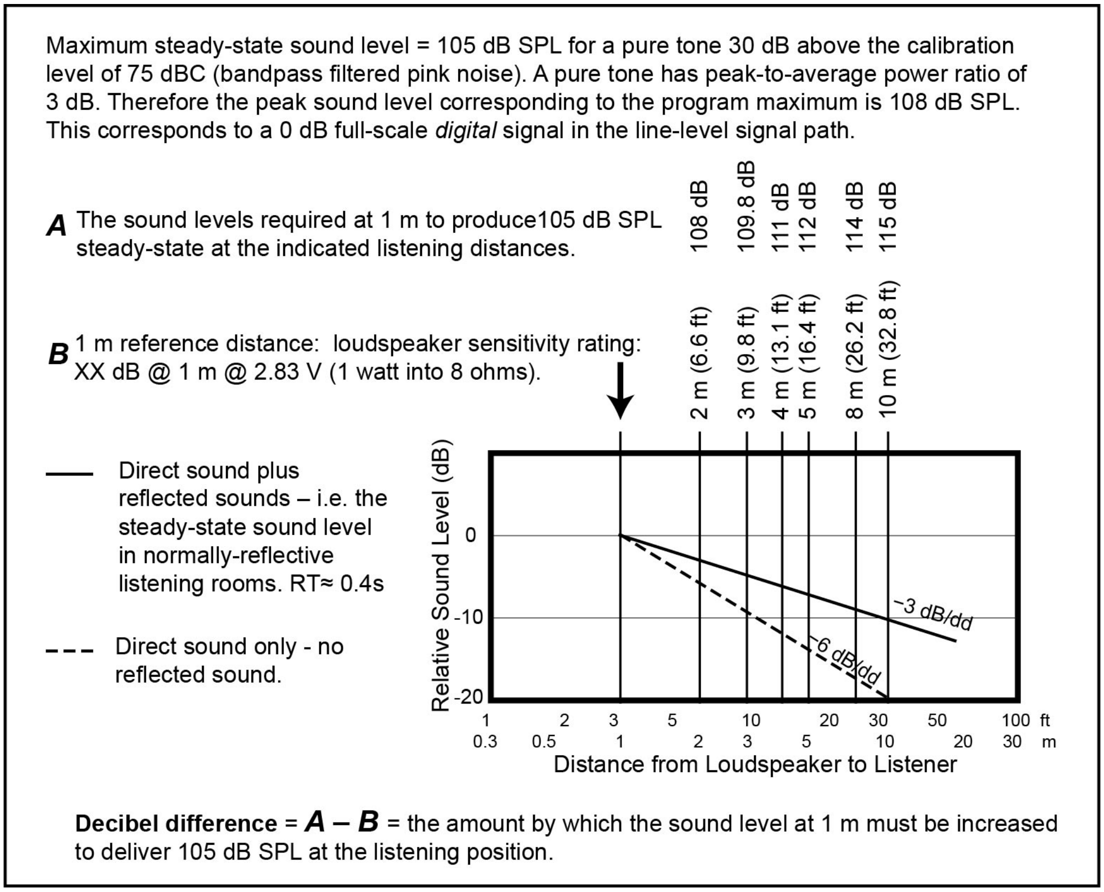


Figure 6. Determining how much the output of the loudspeaker must be elevated above the sensitivity rating (XX dB @ 1 m @ 2.83 V [1 watt into 8 ohms]) to deliver the required maximum sound level at the listening distance relevant for the installation in question—the Decibel Difference. The distance scale is logarithmic, compared to the linear scale in Figure 5.

The following explanation uses a real system as the example.

Step 1: Measure the distance from the front-center loudspeaker to the head location for the prime listener. Using the graph in Figure 6 find the sound level (***A***) that is required at 1 m for the –3 dB/dd curve: direct sound plus reflected sounds. If the distance you measure is not shown you can interpolate.

*Example: The prime listener is about 13 feet (4 m) from the center loudspeaker, so the loudspeaker needs to radiate **111 dB** measured at 1 m in order for the steady-state sound level at my ears 4 m away to be 105 dB SPL.*

Step 2: Loudspeaker sensitivity. Using the guidance in Section 2, above, try to arrive at a trustworthy number for the loudspeaker sensitivity. If the user wishes to make the measurement there are some important cautions. The measurement must be made in a reflection-free environment, or employing a time-windowed FFT measurement that eliminates reflections. Although the specification is at a distance of 1 m, in reality the microphone should be in or close to the acoustical “far field” (Figure 10.9, p. 296). It is common for the measurement to be made at 2 m and calculated for 1 m by adding 6 dB.

*Example: The loudspeakers are specified at **86 dB** @ 1 m for 2.83 V input; measured in an anechoic chamber. Two trustworthy reviews published sensitivity ratings that were very close, so it seems like a good number. This is an average-sensitivity for a cone and dome loudspeaker.*

Step 3: Now, assuming that a realistic sensitivity rating can be found, subtract it from the number determined in Step 1, giving the dB increase in sound level at 1 m that is required to generate 105 dB SPL at the prime listening location; this is called the “decibel difference” in Figure 6.

*Example: Subtract 86 from 111 giving us **25 dB** as the “decibel difference.”*

The following two steps exist in two versions. The first involves calculations. The second is a graphical equivalent.

Step 4: Use the following equation to calculate the amplifier power needed to drive an 8-ohm loudspeaker:

Amplifier power (reflections included) = $10^{\text{dB difference}/10} = 10^{25/10}$ = **316 watts (into 8 ohms)**

Step 5: So far we have simplistically assumed that the loudspeakers have 8-ohm impedances. If that is the reality, no more needs to be done. However, in most cases the real minimum impedance is lower, closer to 4 ohms, so the numbers we have just generated need to be adjusted. The loudspeaker sensitivity rating used 2.83 volts as an input. If the loudspeaker had an impedance of 8 ohms, the input power would be 1 watt ($V^2/R = 2.83^2/8 = 8/8 = 1$ watt). If the loudspeaker impedance were 6 ohms, the power input would be 8/6 = 1.33 watts, and at 4 ohms the power input would be 8/4 = 2 watts. If the minimum impedance dips to 3 ohms, as some do, the input power would be 8/3 = 2.7 watts, and so on.

*Example: The loudspeakers to have a “nominal” impedance of 6 ohms, but a minimum impedance of 4 ohms. The reality is that the impedance curve wanders over a range of impedances—as do all loudspeakers—so the “nominal” number is just that: a single number estimate to describe a much more complicated reality. The good news is that the minimum impedance has been noted, because this is an important indicator of the current demands from the power amplifier. Looking at the measured impedance curves published in reviews it is clear that the loudspeaker is close to 4 ohms over much of the frequency range. **4 ohms** was chosen as the functional impedance for purposes of determining amplifier power needs.*

The power delivered into 4 ohms is 2 × the power that would be delivered into 8 ohms so the real power demand at the 4 m (13.1 ft) listening distance is:

Amplifier power (reflections included) = 316 watts × 2 = 632 watts into 4 ohms.

Suddenly the already large numbers become very large, and many of the amplifiers available for use begin to look very small.

Listening distance makes a significant difference. If the prime listening location were 1 m closer, at 3 m (9.8 ft) the corresponding number would be:

Amplifier power (reflections included) = 240 watts (8 ohms), 480 watts (4 ohms)

A graphical alternative to Steps 4 and 5:

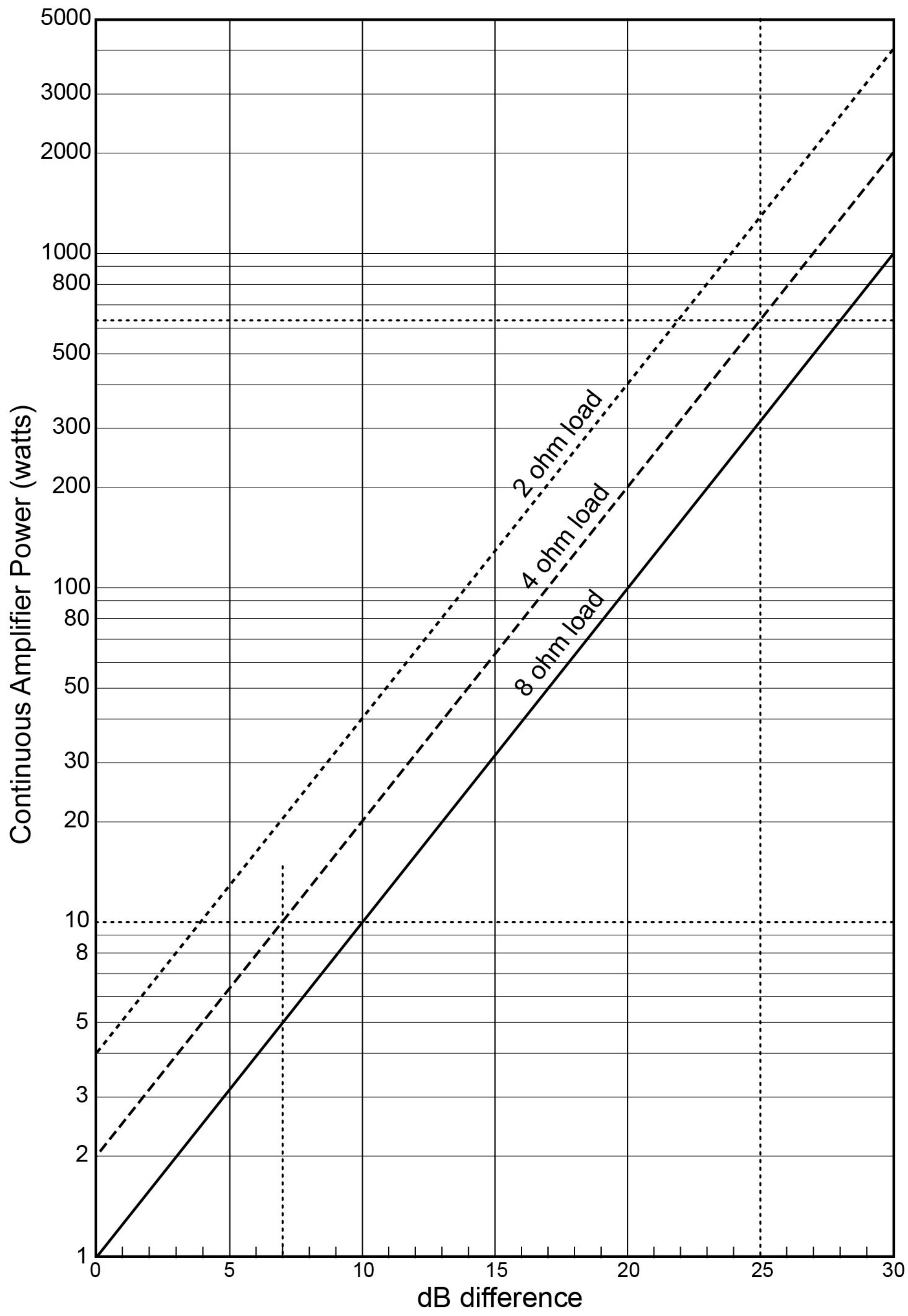


Figure 7. A graphical replacement for Steps 4 and 5. Enter the "decibel difference" from Step 3 on the bottom axis and read the corresponding continuous amplifier power requirement from the vertical axis. Here the fine dotted lines correspond to examples in the text. In one, a dB difference of 25 dB yields an amplifier power rating of 632 watts into 4 ohms. In the other, a dB difference of 7 dB yields an amplifier power rating of 10 watts into 4 ohms.

Example: Does it work?

Power amplifiers rated at 800 watts into 4 ohms (continuous) were installed in the example system. With the volume control set to "0," this system regularly registers C-weighted fast-response sound levels of 106 dB, and peak/impulse levels of 113 dB with routine action movies in a 7.1 configuration. The multiple channels simultaneously active contribute to the very high sound levels measured in the room. A few other movies are allegedly louder than those I experienced. I do not know if the maximum reference levels were achieved in any channel, as that would require monitoring the peak voltage outputs of the power amplifiers. In any event, I found the sound levels to be uncomfortably high. With age it is common to find high sound levels to be unpleasant, even painful (hyperacusis).

Along with many listeners at home, I often turn the volume down a few dB. A reduction of 3 dB is a small change in perceived loudness but it reduces the power requirement by a factor of 2. Suddenly 632 watts becomes 316 watts and everything in the system "relaxes." 6 dB is a factor of 4: 158 watts.

Wishing to preserve my hearing as long as possible, at loud cinemas and rock concerts I wear custom earmold musicians' earplugs (etymotic.com, the inventors) that reduce the volume without destroying the sound quality.

As discussed in Chapter 17, p. 441, hearing damage takes many forms, an elevated hearing threshold being just one indicator. The prospect of hearing less detail in the timbre and space of music and movies is unappealing, and hearing aids are something everyone should try to avoid.

The power-handling capacities of the loudspeakers are yet another parameter that must also be carefully considered. This is another specification for which there are no trustworthy standards. Many people discover the limits of their loudspeakers only when some of the drivers go quiet.

7 Cinema Loudspeakers for Home Theaters

Given the impressive amount of amplifier power required to drive the preceding cone and dome loudspeakers to cinema reference sound levels it is not surprising that some enthusiasts have gone to the professional audio catalogs and purchased cinema loudspeakers having much higher sensitivities. The popular choices include loudspeakers designed for small cinemas, with sensitivity ratings in the range of 100 to 104 dB @ 1 m @ 2.83 V making them substantially easier to drive, even with the common 4-ohm nominal impedance for the two-woofer bass bins.

Let us take the preceding example in which 111 dB SPL was required at 1 m to deliver 105 dB at the listening position 4 m away. The "dB difference" for a 100-dB sensitivity loudspeaker would be 111 – 100 = 11 dB, which yields a need for 25 watts into 4 ohms. The 104-dB sensitivity loudspeaker gives a dB difference of 111 – 104 = 7 dB, requiring only 10 watts into 4 ohms. See Figure 7.

These are easily achievable amounts of amplifier power, so long as the amplifier is able to drive 4 ohms—a specific caution for receiver owners. As for the loudspeakers, with no requirement to be visually attractive and large size not being a disadvantage in cinemas, high efficiency is easily achieved at modest cost.

The question is: are there any compromises when such loudspeakers are used in home theaters? Cinema loudspeakers are designed to entertain hundreds of people in large venues. The installations are calibrated in-situ to deliver X-curve-weighted sound to audiences. As discussed in Chapter 11, the X-curve is not the perceptually optimum curve and, in addition, the traditional calibration process does not deliver consistent sound quality. The prospects for good sound are much better in home theaters.

The requirements for high sound output and highly controlled directivity result in physically large systems. Narrow dispersion (typically 90° horizontal by 40° to 50° vertical) ensures that the bulk of the radiated sound is delivered to the audience, which maximizes efficiency, and avoids reflections from walls and ceiling. This is logical for cinemas but it runs contrary to common practices in domestic listening rooms and home theaters in which listening to stereo music is an option. Most highly rated domestic loudspeakers have relatively wide and uniform dispersion. This is clearly shown in Figure 10.15, p. 302.

As discussed in Chapter 7, this is also a matter of personal opinion, experience and taste. These cinema loudspeakers will clearly appeal to listeners who prefer being in a dominant direct sound field at middle and high frequencies, but will be disappointing to those who prefer the spatial presentation offered by loudspeakers with wide dispersion, up to and including bidirectional-in-phase (bipole) and horizontally omnidirectional designs. A factor in these preferences is the choice of musical genre.

Listeners who employ multichannel upmixing in stereo music will find the dominant direct sound from the front loudspeakers to be less of a factor.

The large diaphragm compression drivers in many of these systems lack the smoothness and high-frequency extension commonly seen in better cone and dome systems, although beryllium diaphragms are better than traditional materials. For movie reproduction, such loudspeakers can be satisfying, and they are more than capable of achieving reference cinema sound levels in small rooms. A few enthusiasts boast about achieving even higher sound levels, which suggests that they are unaware of the hazards to hearing (Chapter 17). These professional loudspeakers have "headroom" that is not needed, and indeed should not be used in home theaters if hearing preservation is a priority.

Achieving the highest performance will likely require some equalization because these systems have been optimized for audiences several meters away in cinemas. As discussed at several points in the book (e.g. Section 12.2.3, p. 348, and 13.2.3, p. 371), the starting point should be to equalize the anechoic—real or FFT time-windowed—on-axis and/or listening window frequency response—explained in Chapter 14. To be effective this should employ high resolution (~ 1/20-octave) data and parametric filters (see Chapter 5). Steady-state in-room measurements, spatially averaged or not, can convey ambiguous information about sound quality.

A recent design that bridges the gap between audiophile cone and dome systems and large cone and horn cinema systems is the JBL Pro M2 monitor loudspeaker (Figure 5.12, p. 132, and the cover of the paperback edition). It competes in sound quality/timbral accuracy with the best audiophile cone and dome systems, with moderate dispersion characteristics (120° × 100°), and has no problem achieving cinema reference levels in home theaters. See Figure 10.15, p. 302 for a helpful comparison of directivities. It is an active system, requiring outboard DSP capability to achieve its impressive anechoic performance.

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