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Small Room Acoustics De-Mythologized

by
Glenn D. White

Glenn D. White is an expert on sound, acoustics and vibration who has been involved with instrumentation of all three for over 30 years. At the University of Washington, he as taught courses in recording techniques, acoustics, and instrumentation. He is the author of the "Audio Dictionary " now in its second edition.

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Small Room Acoustics De-Mythologized

It is certainly no surprise that there are great differences in the perceived sonic qualities of small listening rooms compared to large auditoriums. But it is not so easy to physically characterize, or even accurately describe, what these differences are. And once these differences are more completely understood, how can this understanding help the serious recorded music listener to optimize his listening environment? Several questions come to mind: Can a small listening room be made to sound like a concert hall? Should the listening room be "dead" or "live"? Can the listening room acoustics improve on the acoustics of the original recording? Where should the loudspeakers be placed? Where should the listener sit? Would electrical equalization benefit the sound system? If so, how should this equalization be adjusted for optimum effect?

In an attempt to answer these questions, we will first study the behavior of sound in real rooms and find out how reverberation, echoes, standing waves, resonances, and diffusion come about. We will look into the question of how these objective characteristics of a sound field translate into subjective impressions with which we are familiar. Then we will look at how the architecture and materials of which a room is made effect the objective, and thus subjective acoustics. Next, we will consider the medium of recorded music and will investigate the philosophical differences in various styles of recording and music. Finally, we will try to rationally apply this information to help in the design, or improvement, of a listening environment for recorded music.

Sound Waves, Sound Pressure, and Sound Power

Sound existing at a point in space can be defined as a variation in atmospheric pressure at that point. A sound source, such as a small pulsating sphere surrounded by air, radiates energy into the air in all directions equally, and this is defined as a free field condition. See figure 1. The source S radiates sound energy at a rate called the sound power, measured in acoustic watts. Sound power is defined as energy per unit time, just as any power is defined. The source S causes an atmospheric pressure variation to occur at point P1, at a distance of one unit from the source. This pressure variation is defined as the sound pressure

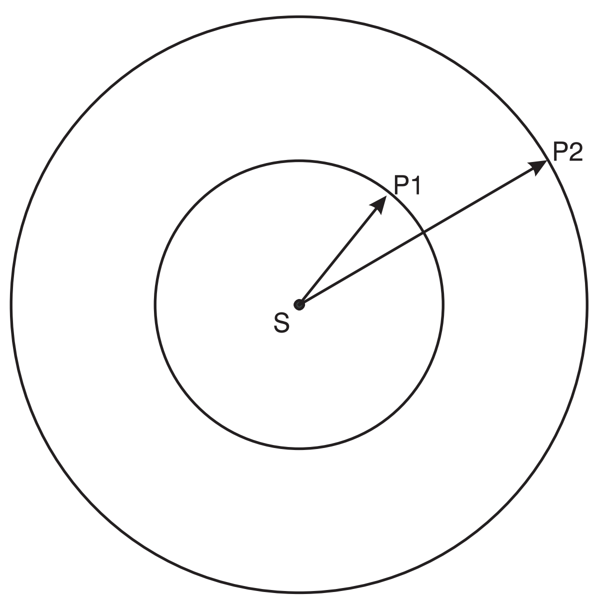


Figure 1

at that point. All points on an imaginary sphere around the source at this same distance will have the same sound pressure. The energy from the source is spreading out in space over ever expanding spheres at the speed of sound, or about 11 feet per second. A short time later, the sound will have reached point P2 at a distance of 2 units from the source. At this time, the energy has spread uniformly on a sphere twice as large as was the case at point P1. The area of this larger sphere is 4 times as large as the area of the smaller sphere, so its strength, or its intensity, will be one-fourth as much. In other words, for each doubling of distance from a source, the sound pressure falls by a factor of 4. This is a consequence of the geometry of space, and is called the Inverse Square Law. Light and radio wave radiation also obey it.

Because the speed of sound is not infinite, each frequency of pressure variation will be associated with a wavelength. If the speed of sound is S, the frequency of the sound is F, and the wavelength is W, then they are related by this equation:

$$S = FW$$

Thus, the higher the frequency, the shorter the wavelength and vice versa. Figure 2 plots the approximate frequency and wavelength of audible sounds. Note that low frequency have quite long wavelengths, in many cases longer than the listening rooms in which we want to hear these sounds. Note also that the highest frequencies have very short wave-lengths. These values are very important, as we shall see shortly

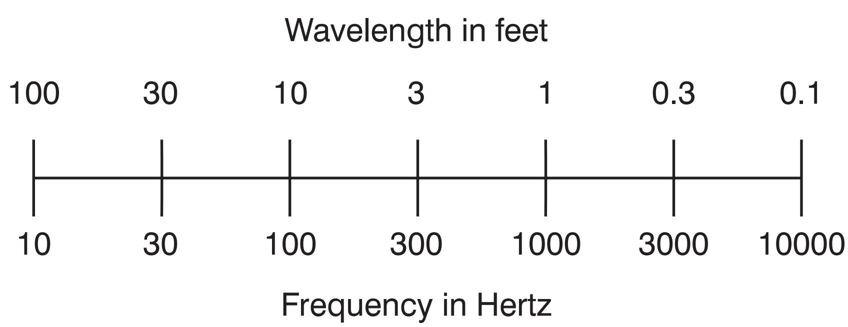


Figure 2

The Behavior of Sound in an Enclosure

For the moment, we will forget about wavelength and frequency, and consider sound in a room from an energy flow standpoint.

A sound source in a room generates a sound field in the room which varies in character at different places in the room, and it is convenient to define several descriptors of this sound field. Locations near the sound source are said to be in the near field of the source. In the near field, the sound is wholly determined by the source itself. A little farther from the source, the sound field begins to be effected by the medium and surroundings. The part of the sound field outside the near field is called the far field. Also, the far field is conveniently divided into regions called the free field and the reverberant field. (There is another part of the sound field, very close to the room boundaries, which is sometimes called the far-out field.)

Before we look at the differences between these parts of the sound field in a room we need to define a quantity called the sound pressure level, or SPL. To do this, we have to introduce the concept of the decibel, or dB. It is quite likely that the dB has caused more confusion in people's minds than any other parameter. The following treatment of it is necessarily brief and not comprehensive, but should provide a working knowledge of SPL measured in dB. Of course, decibels are used in acoustics in many other ways as well.

As we have seen, the sound pressure is defined as the variation in atmospheric pressure caused by the existence of a sound. The sound pressure level is defined as the sound pressure measured as the number of decibels (dB) above 20 micropascals. The dB(SPL) is defined as 20 times the common logarithm of the sound pressure ratio between two sounds. The Pascal is the metric unit of pressure, and 20 micropascals represents the smallest sound pressure that an average healthy human ear can hear at 1000 Hz.

Simply put, the decibel scale is proportional to the logarithm of the actual sound pressure, so equal numbers of dB represent equal ratios of sound pressures. For instance, a doubling of sound pressure is represented as a 6 dB increase, and a four-fold increase in sound pressure is a 12 dB increase. One dB(SPL) represents about a 12 percent change in sound pressure, and is close to the smallest change in pressure that a human ear can detect. These are good rules of thumb which will be helpful if committed to memory. Because of the complex and very non-linear way in which sound is processed in our hearing mechanism, a six dB increase in level does not cause a doubling of the loudness of a sound. It takes an increase of almost 10 dB to double the loudness of a sound. Incidentally, SPL is the quantity measured by all standard sound level meters.

From this, we see that the inverse square law can be described simply by saying that each doubling of distance from a sound source will result in a 6 dB decrease in SPL, provided the source is radiating into a free field, with no nearby reflectors of sound.

Getting back to the sound field in a room (figure 3), the near field is that part of the sound field which exists in the immediate vicinity of a sound source and it is the space where sound is not radiating out in spherical waves. Its extent is usually about equal to the longest dimension of the source but this varies widely depending on the geometry of the source. It may be less than this for low frequencies. The extent of near field is frequency dependent, but in a complex way. It is generally impossible to make meaningful SPL measurements in the near field, except in special cases such as certain loudspeakers which are designed to simulate point sources.

The free field exists farther from the source, and is defined as that space where the inverse

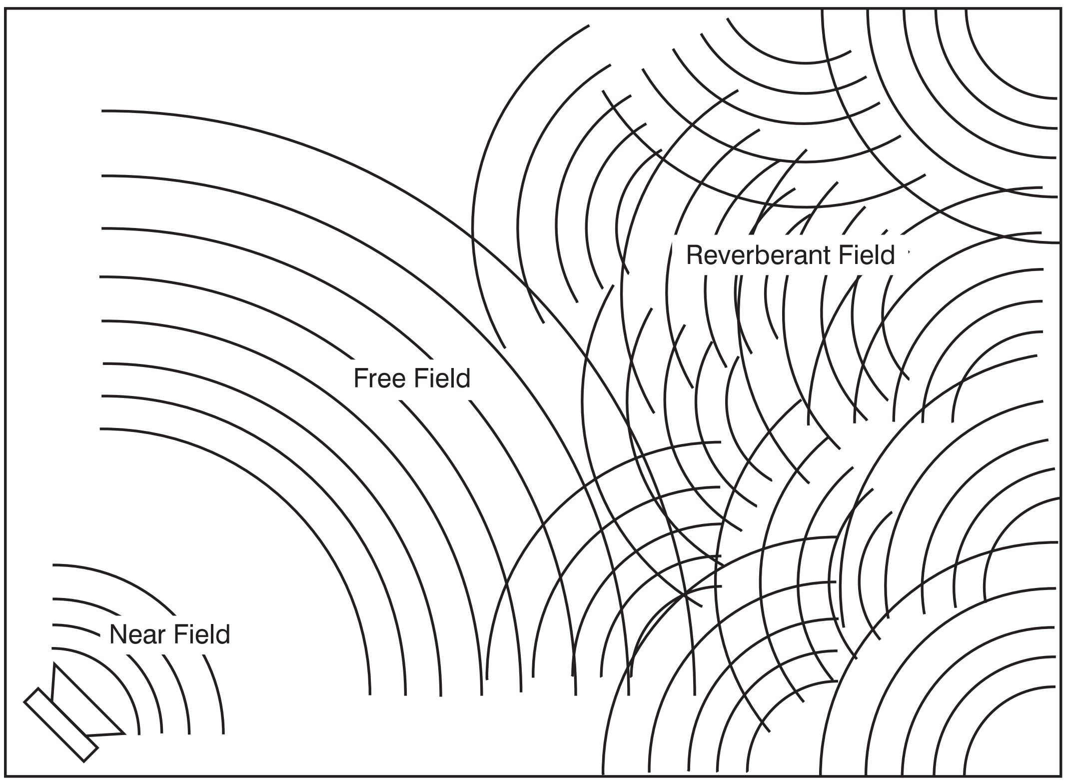


Figure 3

square law is obeyed, at least within acceptable limits. Thus in the free field, the SPL will decrease by 6 dB for each doubling of distance from the source. The sounds reflected from the walls are not adding significantly to the measured SPL, and meaningful and repeatable SPL measurements can be made. This is the region where almost all loudspeaker frequency response measurements are carried out. The extent of the free field is generally highly frequency dependent, always being farther at higher frequencies. The free field is also the best place for a listener to position himself for listening to music reproduced via loudspeakers.

It is important to realize that the free field around a loudspeaker depends greatly on the directional characteristics of the loudspeaker. A highly directional speaker such as a high-frequency horn will have a larger free field than will a more nearly omnidirectional speaker such as a dome tweeter or a small woofer.

As we go farther from the source beyond the free field, we enter the reverberant field, where the sound is more diffuse and homogeneous rather than coming from a particular direction. The SPL in a reverberant field is constant at varying distances from the source. In a true reverberant field the net energy flow through any point is zero; i.e. the energy is arriving randomly and uniformly from all directions at once, providing no clue as to where the source of the sound is located. A perfectly diffuse sound field is not possible to achieve in practice because of such disturbing factors as stationary, or "standing" waves, as we shall see.

The boundary between the free and reverberant field around a loudspeaker is called the "critical distance", and varies with frequency and the directionality of the speaker. It is the place where the direct sound from the speaker and the reverberant sound are equally strong.

It is possible in some rooms to have a condition where there is no free field at all. The reverberant field is dominant all the way up to the near field of the source. This is called a reverberation chamber, and is used for special test purposes - never for listening to music. It is also possible to build a room which has essentially no reverberant field, and the free field extends all the way to the room boundaries. This is called an "anechoic chamber", and also is used for testing, but not for music listening.

If the walls of a room are quite reflective, the sound pressure will increase close to the boundaries. This is because the incident pressure adds to the reflected pressure, and in the case of a perfect reflector, the pressure at the surface would be double the free field pressure at that point. In other words, the SPL will be 6 dB higher at the surface of a perfect reflector. The pressure increase extends out from the wall almost a half wave-length, so this effect is also frequency dependent. This part of the sound field is sometimes called the far-out field, and is not generally used for making SPL measurements.

In any room, the reverberation time affects the location of the free and reverberant fields. Reverberation time is defined as the time it takes for a sound to decay by 60 dB after the source of the sound is cut off. This sounds on the surface like a simple measurement, but it can be highly complex. The only situation where reverberation behaves in a predictable manner is where the reverberant field is truly diffuse, or almost completely random in its direction. Real rooms suffer from lack of diffusion due to the existence of standing waves and "room resonances". These supplementary resonances distort the ideal shape of the reverberation level versus time curve, which would otherwise be a smoothly decaying straight line. It is these resonances that complicate the art and science of loudspeaker placement and equalization.

Standing Waves and Room Resonances

Standing waves arise when two reflective surfaces on opposite sides of a room are parallel and a sound source is between them. At certain well-defined frequencies, the incident sound wave at a surface will reflect back and interfere with itself causing a stationary pattern of low and high sound pressure levels. See figure 4.

With perfectly reflecting walls, the minima would be zero sound pressure, or negative infinity dB SPL, but ordinary walls do not cause complete cancellation, and the minima are only down about 10 to 30 dB. Note that a set of standing waves will exist which are integral multiples of the frequency where one half wavelength is equal to the wall spacing. The behavior of parallel walls is similar to what happens in an organ pipe. These standing waves are also known as room resonances. A good familiar example of a strong standing wave is the tiled shower stall. If the height is 7 feet, the lowest frequency which will be reinforced at the ceiling and floor is about 80 Hz, which is within the fundamental frequency range of most male voices. The nearby side walls keep the sound energy confined in the stall, increasing the loudness, and the position of the head near the ceiling means the pressure increase due to the standing wave is heard. This strong reinforcement of the lower voice range is a powerful inducement for some people to sing!

A rectangular room with smooth hard surfaces will have three sets of standing waves, and if the dimensions of the room are equal (a cubical room), all the resonances will overlap and reinforce each other. If the dimensions are commensurate (having a least common denominator), then some of the resonances of the three sets will have the same frequencies, and will reinforce each other also. This increases the non-uniformity of distribution of sound in the room, and for this reason non-commensurate dimensions are desired for music listening rooms.

Note from figure 4 that all standing wave patterns have maxima at the room boundaries. Note also that if one listens at any location where there is a minima, that particular standing wave will not be heard, or at least will be heard at a much lower level.

Standing waves in rooms are a form of room resonance, and their location and frequency are precisely determined by the room dimensions, not by the source of the sound. Their “strength”, or magnitude depends on the sound absorption characteristics of the room boundaries. If the boundaries are very absorbent, the standing waves will be very weak, and probably not noticeable at all. On the other hand, if the boundaries are very reflective, such as concrete or smooth plaster, the standing waves will be very prominent. Most common surfaces in music listening rooms have significant absorption, especially at the higher frequencies. For this reason, standing waves at high frequencies are seldom if ever a problem. At low frequencies, however, two things conspire to make standing waves very annoying to the music listener. One is the fact that the wavelengths are fairly long. For instance, at 100 Hz, the wavelength is about 10 feet, and standing wave maxima are thus about 5 feet apart. Contrast this to 10 kHz, whose wavelength is about an inch, with standing wave maxima spaced only about a half-inch apart. The widely-space maxima and minima of low-frequency standing waves are what give a small bad room its non uniformity.

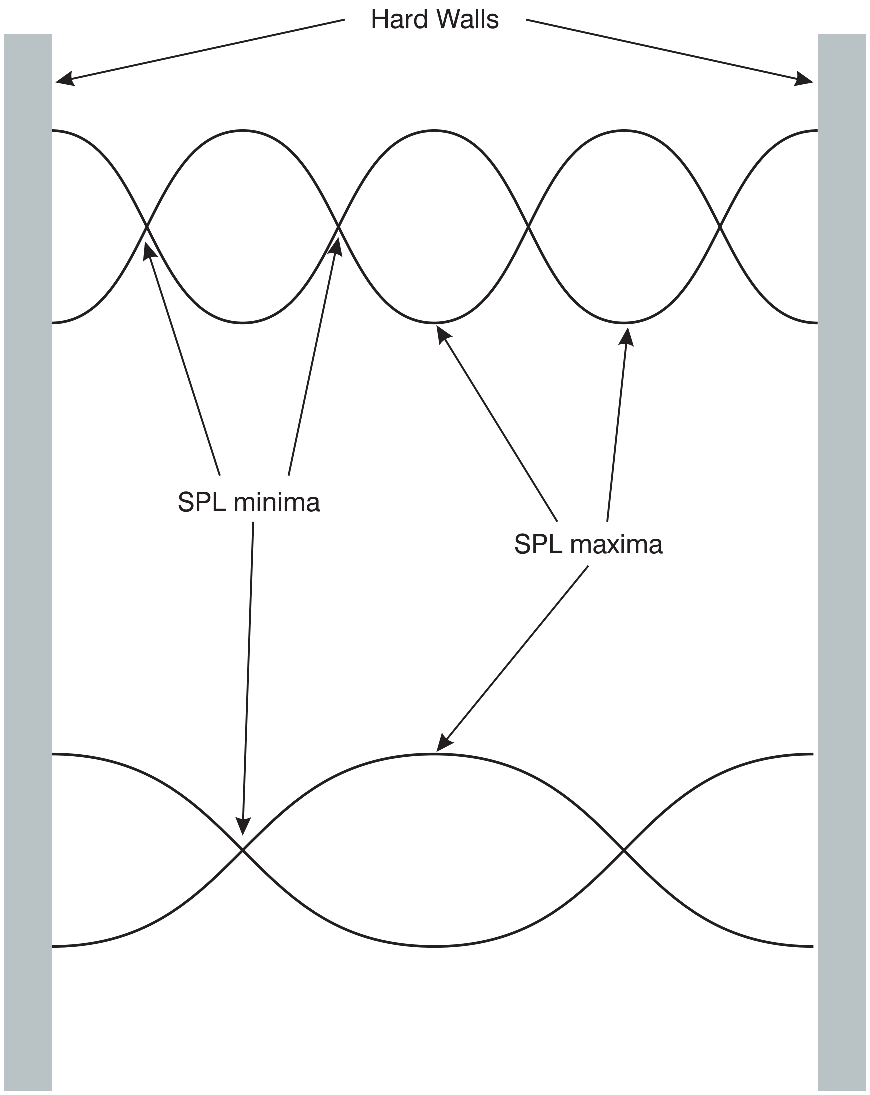


Figure 4 - Standing waves between two parallel surfaces

Now we can begin to define the difference between large and small rooms. In a large room (smallest dimension of 50 feet or so), the standing waves will be multiples of very low frequencies. Two walls 50 feet apart will have a lowest resonant frequency of about 11 Hz, and standing waves can exist at all multiples of this frequency. they are thus very close together by the time they get up into the audible range, and even though they are extremely numerous, they tend to simply blend together fairly smoothly. This is the reason large rooms always sound smoother and more uniform than small rooms. Of course, large rooms can have plenty of problems of their own. For example, another acoustical problem caused by parallel surfaces is the phenomenon of “flutter echoes”. If the spacing between the surfaces is quite large, usually about 25 feet or more, a sharp transient sound produced between them will reflect off one and then the other giving a series of echoes which gradually die away. This can be annoying, and usually is worse in fairly large rooms.

There are other acoustical problems associated with rooms besides standing waves. For instance, the corner where two walls and a floor or ceiling meet (geometrically, this is called a “right tricorner”), will reflect sound of any frequency directly back in the direction from whence it arrived. (This is the principle used in radar reflectors and also the red light reflectors found on many bicycles and cars.) Two right tricorners in a room, especially if they are diagonally opposite each other, will give rise to standing waves just like parallel surfaces will. Also, in large rooms, a right tricorner will give an echo, regardless of the location of the sound source. If the angles meeting at the corner are not right angles, the tricorner does not reflect back in the same direction, and therefore architects are always admonished to avoid right tricorners in music rooms, regardless of their size.

Also, because of the huge numbers of possible standing wave patterns in large rooms, they tend to have a much greater degree of diffusion in the reverberant field than do small rooms with their fewer and more widely-spaced standing wave frequencies. This diffusion lends a sense of “envelopement” in the sound field, and also provides a sensation of smoothness and uniformity to music. To achieve this in a small room is nearly impossible, for if the absorption by the surfaces is sufficient to prevent standing waves from becoming objectionable, the reverberation time will be so short that the room will sound “dead”, and live music will not be appreciated, especially by the musicians. In order to have some sound diffusion in its reverberation, small rooms should not have parallel surfaces or right tricorners, and this can be attained in specially designed music rooms. But a living room in a residence is usually another matter, as is quite obvious.

Types of Recordings

Before we look at the effect of reproduced music in small rooms, it will be instructive to consider the characteristics of various types of recorded music. Consider a recording of a large orchestra. Typically the recording will have been made in a large auditorium, one presumably selected for its fine acoustics. Such a room will have significant reverberation, and at least some of this reverberation will be recorded along with the direct sound from the orchestra. How much “room sound” is captured by the recording is of course up to the recording engineer, but in any case it will be significant. The intent of such a recording is to provide the listener with the illusion of being in the large room with the orchestra. It is not intended to try to re-create the orchestra in the listener’s living room. For this type of recording, it can be argued that a dead listening room is desired, for the correct acoustical ambience is in the recording, and should not be compromised by the addition of the ambience of the listening room, for the listening room acoustics will always be less desirable as we have seen.

On the other hand, a recording of a very small ensemble, or in the extreme a soft solo instrument such as a Spanish guitar, need not have the ambience of a vast hall recorded with it. To preserve the intimacy inherent in the guitar's music requires relatively close mike placement and the necessary reduced amount of the recording room environment. (However, it is a grave mistake to make such a recording in a small reverberant room, for the boundary reflections will be relatively strong and will be recorded, even with fairly close miking. In general, the larger the recording room, the better.) This recording, if reproduced in a small dead room will also sound lifeless, and some reflected sound is desirable at the listening position. It can probably be said that no one room will be ideal for listening to large orchestras and small musical groups; one has to compromise in one direction or the other.

It should be mentioned in passing that the complex sound systems which emulate large rooms by adding reverberation and supplying synthetic side wall reflections from supplementary speakers require non-reverberant listening rooms. What a listener hears is always the summation of the characteristics of his listening room and the ambience of the sound field provided by the sound system. In no way can any sound system cancel out the acoustics of the listening room.

Having said this, we must hastily point out that it is possible to optimize the listening conditions in a room by seeing to it that the sound system will excite the undesirable characteristics of the room as little as possible.

Frequency Response

The frequency response of a sound system can be defined in several ways. For instance, a system consisting of an amplifier connected to a loudspeaker in an anechoic chamber can be said to have a flat frequency response if the SPL produced at a distance of 1 meter in front of the loudspeaker is uniform at all frequencies when a variable frequency sine wave is input to the amplifier. This is sometimes called the free-field response of the speaker, and many manufacturers measure their speakers this way.

Another way to measure frequency response is to place the speaker in a reverberation chamber and put the measurement microphone in the diffuse or reverberant field. The resulting curve is called the power response of the speaker, and some manufacturers measure loudspeakers this way also. A speaker with flat power response will not necessarily have a flat free-field response, for the power response integrates all the energy produced by the speaker in all directions, while the free-field response measures the sound projected directly forward. If the speaker was completely omnidirectional at all frequencies, its power and free field responses would be the same. (This may or may not be desirable, for an omnidirectional speaker tends to excite all the standing waves and resonances of the room, while a directional speaker theoretically will excite only those resonances which occur in the direction the sound is aimed. In practice, it is difficult if not impossible to control accurately the directivity of a loudspeaker system over a wide frequency range. All systems tend to become omnidirectional at very low frequencies because they are small compared to the wavelengths involved, and to be quite directional at high frequencies because they are large compared to these smaller wavelengths. Referring to figure 5, the directional loudspeaker can represent a horn tweeter or a planar speaker such as an electrostatic. The non-directional speaker could be a dome tweeter, but also represents the situation for low frequencies radiated by almost any woofer. Therefore it is difficult to control poor room acoustics by using directional speakers because most of the resonance problems in rooms occur at low frequencies.)

One might think that a listener would find a flat free-field response to be ideal, and this might be true in most rooms if he listened at one meter from the speaker. This of course is seldom desirable or possible. As the listener moves farther from the speaker, closer to the reverberant field of the room, the perceived frequency response changes, and always for

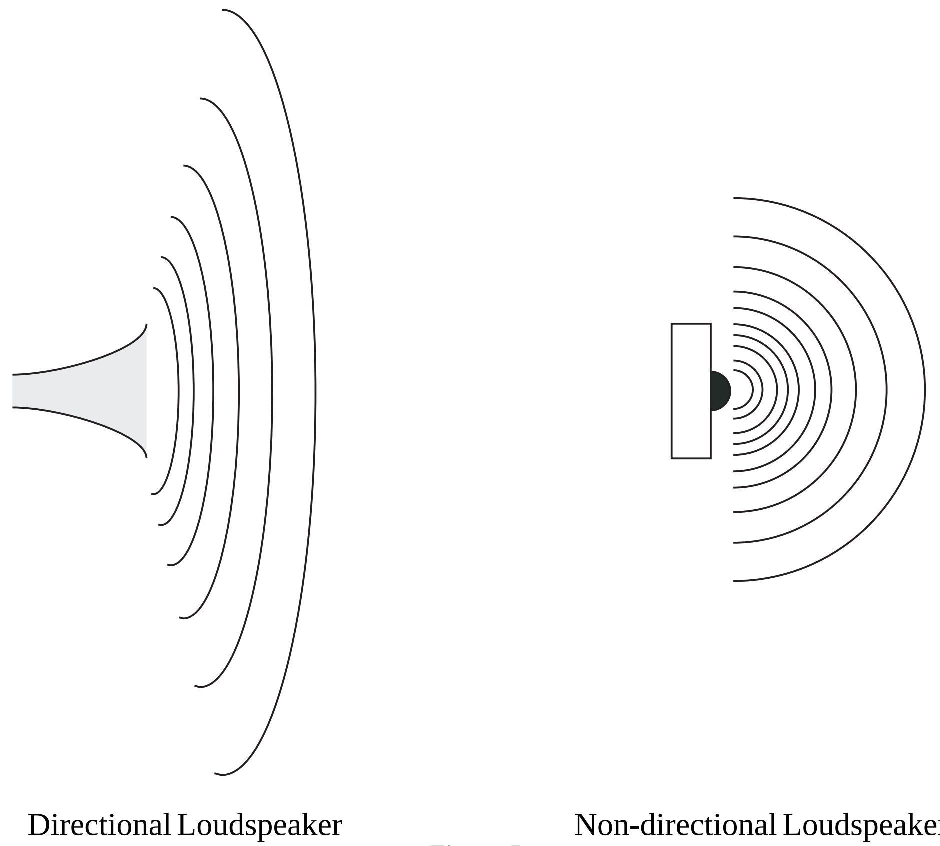


Figure 5

the worse. The speaker will excite various standing waves of the room depending on where it is located. If it is in a room corner, it will of necessity excite all the standing waves. If it is placed out from the walls a few feet and not close to a corner it will selectively excite those standing waves which have maxima near the speaker. (A sound source located at a minima of a standing wave will not excite that particular resonance very much. If the minima were truly a null, it would not be excited at all.) Therefore, the location of the speaker in the room has a great deal to do with how much energy it puts into the room as a function of frequency.

In particular, the proximity to reflecting surfaces greatly affects the efficiency with which low frequencies are radiated. A corner location provides the most low-frequency energy, and a side wall location provides somewhat less, while a location away from walls provides the least. Some loudspeakers are designed to be flat when in a corner or against a wall, while others are designed for mounting away from these surfaces.

Remember, however, the standing wave pattern in any room is determined by the room dimensions only, and does not change if the speaker is moved around. Loudspeaker placement can only affect the relative degree to which various resonances are excited. Also, for any given loudspeaker location, the perceived frequency response depends greatly on the location of the listener. If one is sitting near the maximum of any standing wave, that frequency will be accentuated. It follows that if a speaker is in a corner and the listener is in the diagonally opposite corner, both will be at maxima for all standing waves, and the most efficient perceived low-frequency response will be attained. Such a listening position has many other disadvantages, however!

Attacking The Problem

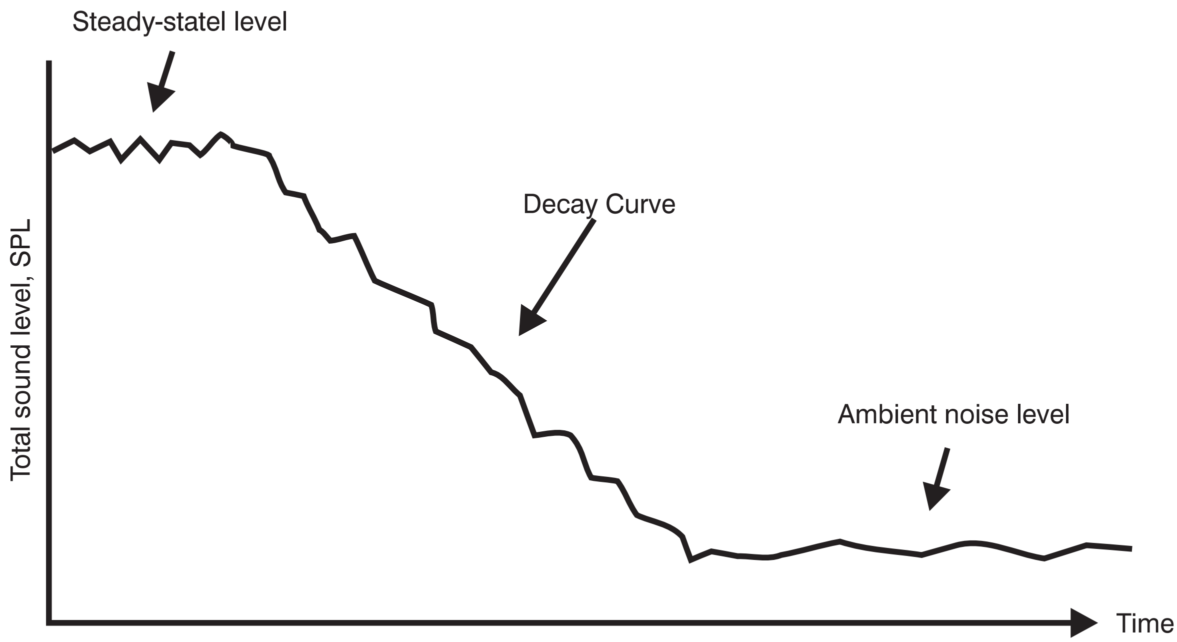
It might seem that any listening situation could be corrected to have ideal frequency response simply by applying corrective electrical equalization in the sound system. This, however, is an oversimplification. The perceived frequency response of a sound system depends strongly on the speaker location and the listener location. For one listener position and speaker position, equalization can provide flat response for steady-state signals.

The term “steady state” is important. Remember that standing waves are formed by reflections back and forth between parallel surfaces. It therefore takes a little time for these standing waves to build up. The listener to a sound system, however, hears the direct sound from the speaker first, before any reflections take place. This direct sound will not be contaminated by standing waves, and if the loudspeaker has flat free-field response, after equalization, the first sound arriving at the listener will be uniform in response. Only later will the overall response take on the coloration caused by the room itself. It so happens that the ear is particularly sensitive to the first sound it hears, and tends to judge a sound’s overall quality by the uniformity of the direct sound. Therefore, even though a large amount of equalization can correct for a very strong maximum or minimum from standing waves, the correction will be applied to the direct sound where it is not needed, and the timbre of the sound may be distorted. For this reason, equalization must be used carefully, and always must be adjusted by listening to the results and not relying exclusively on instruments such as sound level meters to determine “flat” response.

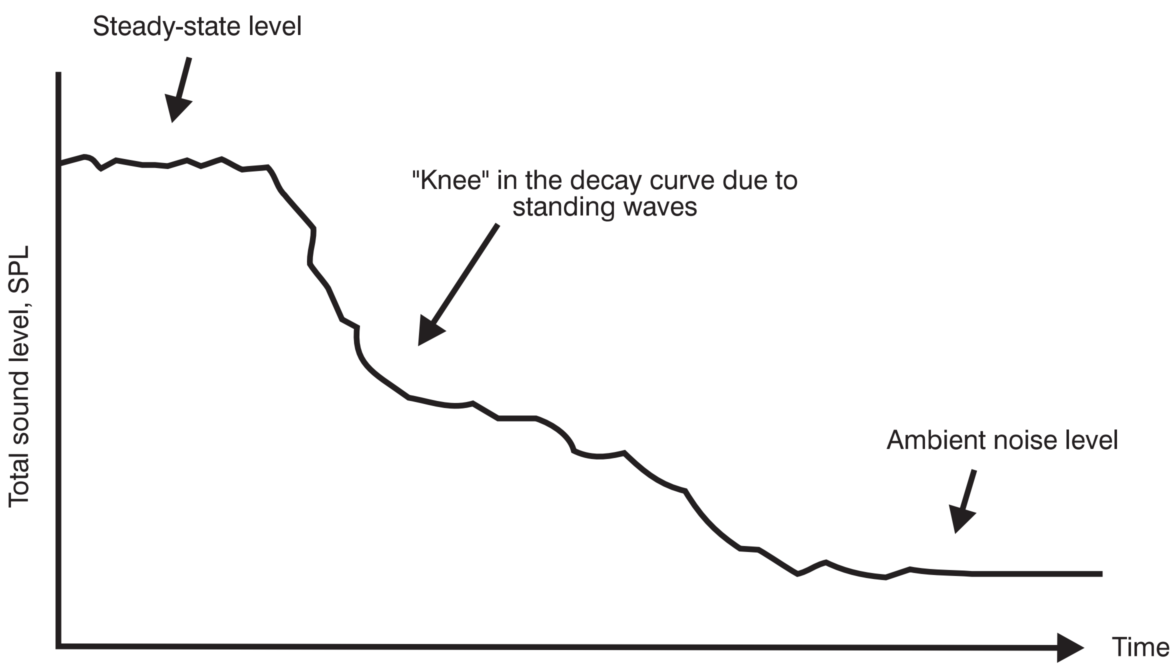
Equalization for Improvement

Probably the best thing to do first to optimize the listening conditions of a sound system is to reduce standing waves and resonances as much as possible. Standing waves in a room will distort the natural shape of the reverberation curve, as in figure 6. Try to eliminate large parallel surfaces. Large pieces of furniture with irregular surfaces such as bookcases can reduce standing waves by adding diffusion to the reflected sounds. Sound absorbing materials or structures can be applied in some cases, although low-frequency absorbers are necessarily large. Flexible walls, such as sheetrock on steel studs are quite good at absorbing low frequencies, especially if backed by damping material such as rock wool insulation. Draperies, if made of dense material, are effective at mid and high frequencies, but are not very good absorbers of very low frequencies. Carpets and overstuffed furniture help prevent the floor and ceiling from resonating. Of course a cathedral or sloping ceiling is much preferred, but not always practical. Avoid placing the listening position near hard reflecting surfaces. Then, experiment with speaker placement while listening at many locations in the room, and try to establish the placement which provides the most uniform low-frequency response, consistent with domestic harmony. The listening position can seldom be one precise location. Try to find a combination of speaker and listener positions that result in the least change in the sound.

After these steps are taken, electrical equalization can be very beneficial. A suitable graphic equalizer will be found to work well. In most rooms with standing wave problems, it will be found that the low frequencies will be the most troublesome, and that several maxima will be quite close together in frequency. At higher frequencies, it will be found that the resonances are blending together into a smoother overall response. For this reason, the equalizer used should have narrower bands (i.e. one-third or one-half octave) at the lowest frequencies and need not be so narrow at mid and high frequencies. The broader the bands, or the more gentle the adjustments that can be used at mid and high frequencies, the better the result the equalization will produce. The response will be more uniform.



Ideal reverberation curve



Actual reverberation curve

Figure 6

A good way to perform the equalization is to use a CD or record with one-third octave bands of pink noise. This is especially convenient if a one-third octave graphic equalizer is in the system. A sound level meter can be used to detect the level. Then, adjust the equalizer for uniform response with the microphone of the sound level meter at the listener position, always using the smallest amount of equalization. The knobs of the graphic equalizer should be as close to the “0” line as possible to avoid ripple in the overall response curve, as illustrated in figure 7. Though the better designed equalizers have less of this ripple. It is important to have a very smooth equalization curve at the mid and high frequencies, without any abrupt peaks or dips. In the low-frequency range, fairly pronounced peaks and dips can be used without noticeable distortion of the direct sound. Don’t worry too much about minima in perceived loudness, for they are much less audible in music listening than peaks, and if you correct an acoustical minima by adding a peak by equalization, the peak will show up strongly in the direct sound as discussed before. It is also important that the highest frequencies should not be boosted to attain flat response at the listener position; otherwise the system will sound excessively bright. The ideal system response curve at the listener position depends on the size of the listening room, but typically should roll off by about 6 dB at 15kHz. Be guided by your ears - they are a more sophisticated measurement tools than any instrument!

In general, electrical equalization, correctly designed and applied, can make a room sound better. It can make a good room sound excellent, but will not make a bad room sound great.

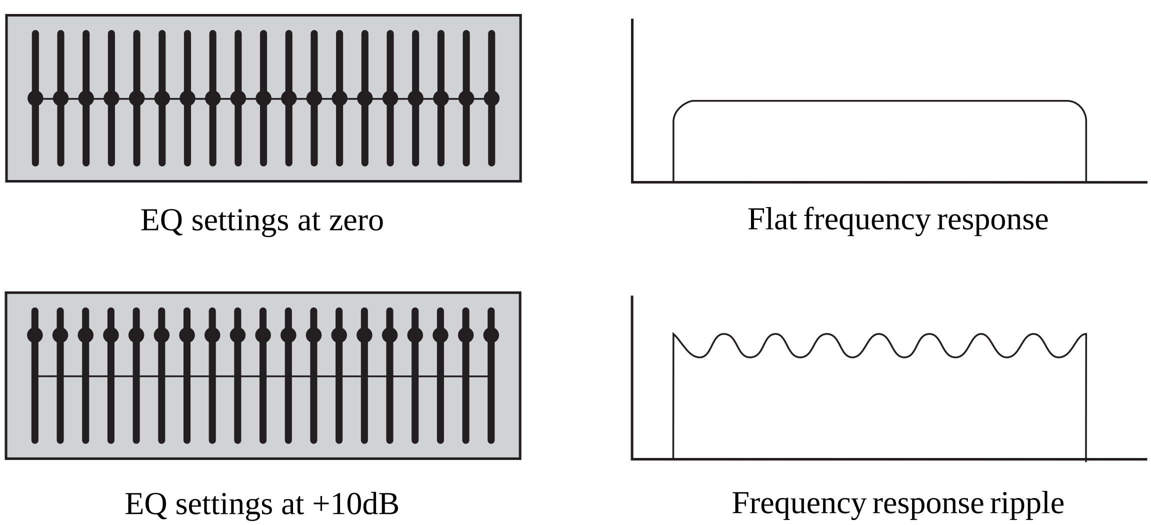


Figure 7