CHAPTER 4 ROOM ACOUSTICS

Room acoustics is an complex subject on which numerous books have been written. However, virtually none of them apply to small rooms. There simply has not been much demand for an understanding of the sound in small spaces. Automobiles are the single most common listening room in the world, but serious listening is just not a high priority. Even in the automobile little work has been done on small room acoustics. In my years working for auto related companies I have both studied and contributed to the state-of-the-art in small room acoustics. Large rooms have large budgets associated with them, and large budgets drive extensive study. The large room acoustic problems are well understood and it is unfortunate that so little of it is applicable to small rooms.

In this chapter, I will talk about the basics of what is known about small rooms, and I will pay particular attention to their differences from large venues. I will not be able to cover this subject in detail (it is a good topic for another book). I will provide you with a reasonable background and some specifics to pay attention to. The detailed theory behind this background information will have to remain transparent for now.

4.1 The Small Room

4.1.a Differences From a Large Room

RT_{60}
the Reverberation Time to reach 60 dB below the excitation level.

In room acoustics, there appears to be a single number which we all believe that we understand and that is the reverberation time. This is the time it takes for the steady state sound level in a room to drop by 60 dB and it is denoted by **RT_{60}**. Many acousticians feel that the –60 dB level is far too low and that we can really only perceive sound decay down to about a 20 to 30 dB reduction. To me this is quite plausible, but in a small room it is also irrelevant. RT_{60} times in small rooms are simply too short to be of significance or to even measure for that matter. The time required to drop to only 20 dB is simply miniscule. Examining why these reverberation times are so low is enlightening.

Mean free path
the average distance that a sound wave travels before it meets with a surface.

An acoustic space is primarily characterized by its volume and its surface area—all spaces have these two parameters clearly defined. As a space gets larger its volume goes up more rapidly than its surface area. This is easy to see for a sphere where the volume goes up as the third power of the radius while the surface area only goes up to the second power. As a result, the ratio of the volume to the surface area goes up linearly with the radius and a large space will have a relatively small surface area when compared to a small space. Since virtually all sound absorption occurs at the boundaries of an enclosure, sound absorbing material placed on the enclosure surfaces is much more effective in a small room than a large one. This can also be seen by considering what is called the **mean free path**—the average distance that a sound wave travels before it strikes a surface. Clearly, the mean free path for a large room is ten to a hundred times greater for the large room than the small room. For example, a patch of sound absorbing material placed on the wall in a small room is "seen" by a sound wave ten to a hundred times as often as this same patch in the larger room. Even if the entire room is covered with this material, it will still be seen ten to a hundred times more often in the smaller room than the larger one. As a general consequence of these facts we can say that small rooms are naturally more dead (low reverberation) than larger rooms since some level of sound absorption is always present and it works to dampen the small room much more effectively.

The mean free path also enters into the situation when we consider the human hearing system. As I stated in Chapter 2, the human hearing mechanism integrates over a time period of about 10–20 ms. The mean free path is usually shorter than this time in a small room. It is virtually impossible to perceive a clear direct sound in small rooms (unless they are **anechoic** chambers). This simple fact accounts for a major portion of the difference between small and large room acoustics. There have been virtually no psychoacoustic studies of sound perception on this short of a scale of time delay. In addition, the fact that there is virtually no chance of receiving a clear direct sound results in some significant implications for small rooms.

Anechoic
a room that is free of any sound refections.

With the lack of a perceivable direct sound component from the loudspeakers, it is important to understand the implications of the steady state sound in rooms. But, in order to understand the steady state, I first need to describe the modal characteristics of a space.

4.1.b Room Modes

Mode
a resonance, but more specifically the shape of the vibrations at that resonance.

It is probably common knowledge that rooms have **modes**—also called resonances. But, I think that there is a real misconception about what the modal behavior of a room means. Figure 4-1 on page 56 shows the simulated frequency response for a room with dimensions of 3 m by 4.2 m by 6 m, a total volume of 76 m^3, which is a large HT, but an extremely small room from a commercial theater standpoint. This room is tightly sealed with a light amount of damping on the walls. The first mode (excluding the static pressure mode at 0 Hz) occurs at about 28 Hz, the next mode occurs at about 40 Hz. As the excitation frequency goes up, the modes get closer and closer in frequency—the **modal density** (the number of modes in a fixed frequency band) increases. Note that the fluctuation of the response curve—the "variance"—decreases up to about 100 Hz. This decrease occurs because the modes carry and sustain the energy in a room, and, the more of them that there are, the smoother the response will be. This holds true only to about 150 Hz after which we see a completely different characteristic evolving. Note also that the mean level is approximately constant up to about 150 Hz after which it appears to drop in level by about 3-5 dB.

Modal density
the number of modes in a given frequency range.

Geometrical Acoustics
The frequency range in a room where the sound waves travels as rays—like light.

The region above about 150 Hz in this room is called the **geometrical acoustics** region. In this frequency region sound travels in a ray like motion, like light, since there are enough modes to carry sound propagation in any

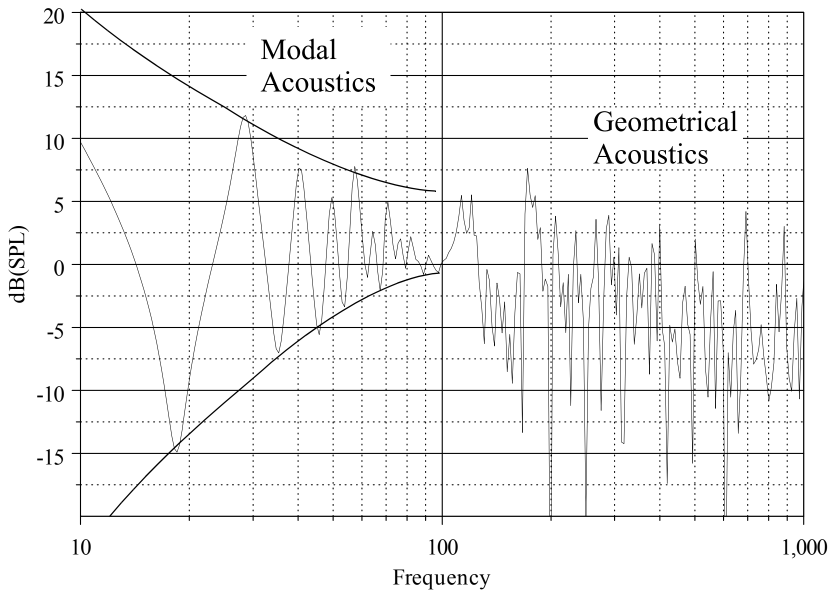

Figure 4-1. Frequency response of a small room in dB (SPL).

direction. Below the 150 Hz point there is a distinct modal characteristic to the room. Modal behavior only occurs at these lowest frequencies. Of note is the fact that this transition frequency (called the **Schroeder Frequency** (f_s) after the German acoustician Manfred Schroeder) moves lower in frequency for larger rooms. In fact, virtually all commercial theaters have a Schroeder frequency which is low enough that any modal effects are all below the audio bandwidth. Commercial theaters and auditoriums simply do not have to consider the modal region of a rooms acoustic sound field.

Schroeder Frequency the frequency at which a room transitions from statistical to geometrical.

4.1.c Modal Chaos

It is important to understand the nature of the sound field in the geometrical acoustics region, above the Schroeder frequency, as this is most of the audio band. This region is sometimes called the **statistical acoustics** region for reasons that I will now elaborate on.

Statistical acoustics the region where the sound field can only be discussed statistically.

Note in Figure 4-1, that above f_s, the nature of the frequency response curve looks pretty much constant, independent of frequency. It has numerous peaks and dips, but these do not appear to get narrower and closer together with increasing frequency as the response below 100 Hz does. In order to examine this feature in more detail, consider the frequency response curve in the previous figure zoomed in to a narrow region between say 700 and 750 Hz (the actual frequency range is not important). I have shown this zoomed in response in Figure 4-2. Note that there are three

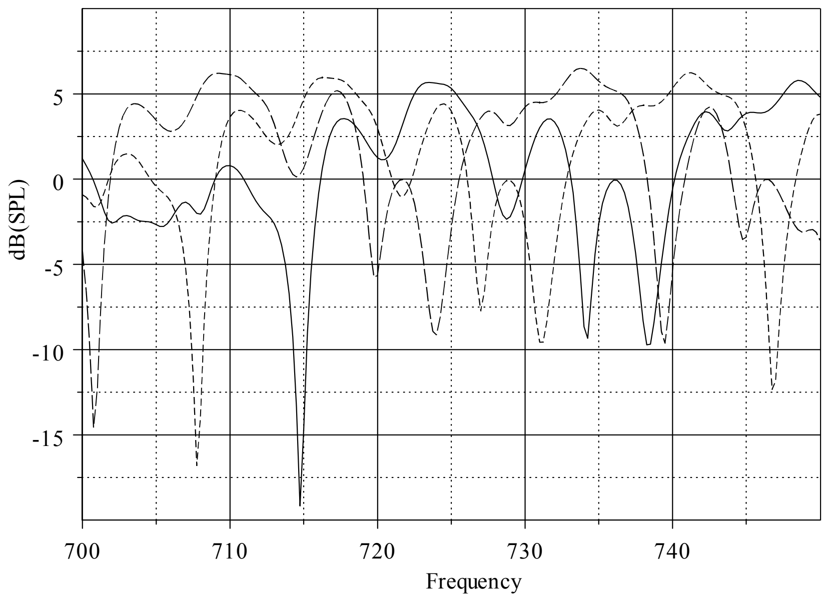

Figure 4-2. Detail of the frequency response in a room at higher frequencies—in dB(SPL).

curves in this figure. These three curves are frequency response curves for the same room, but with a minor perturbation of the wave speed or alternately the measurement location (they amount to basically the same thing). This kind of wave speed perturbation will almost always occur in a real room, since the amount of speed change is so small. A slight temperature change or a small air current or a person in the room shifting listening position slightly, almost anything would cause perturbations of a magnitude sufficient to cause the response variations shown in Figure 4-2. These three curves appear to be completely different, and they are in detail, but as it turns out their statistical properties are very consistent. They all have the same mean and the same standard deviation; statistically each one of these curves can be thought of as a single sample from the same stable (stochastic) random variable.

The statistical properties of the frequency response curve in rooms are well known. The distribution of dB levels is not symmetric about the mean line, the curve lies above the mean 33% of the time and below it 66% of the time. It lies 66% of the time in a dB band about the mean level that is plus 6 dB and minus 11 dB. The really important thing to remember here is not the statistical values themselves, but the fact that the frequency response curve above f_s, is not a deterministic quantity. A single point microphone measurement of a pure tone that is intended to measure the mean response of the room will actually fluctuate about this mean by the amounts that I have noted. Stated another way, if one attempts to measure the mean level of a tone in a room, it will have an expected error of +6 and –11 dB—hardly a useful measurement. Basically, a single point microphone measurement will not result in the actual room response, it is merely one sample of the room's statistical response.

This can be seen in practice on a spectrum analyzer. Excite the room with a pure tone and watch how the level on the analyzer varies in time. This is the effect of these very small perturbations of the frequency response. The wider the bandwidth of the analysis the smaller the level variations will be, but of course, the poorer the resolution of the response.

These considerations are important to note, but I should also point out another important facet of this discussion. These statistics only apply to the steady state response in the room, not to the direct response of the loudspeaker. In many small rooms the direct response from the loudspeaker is usually quite short since the reflections often arrive within a few milliseconds and the room reaches a steady state in little over this time period. The point here is that in a small room there can only be statistical measurements as nothing else makes much sense or is meaningful. This has profound implications to measurements of a small room's acoustics. (Later I will show how it is possible to make meaningful measurements in a small room at very short times, but this takes special processing.)

The standard way of dealing with this issue is the same one that is used for the investigation of any random variable, and that is to take a number of samples and average them, thereby reducing the variability. It turns out that one can achieve this averaging effect in two different ways. The first is to average over frequency, and, if the measurement bandwidth is of sufficient size, then the measurement variability will drop down to something more useful (+6,–11dB variations are not really very useful). However, to get down to a ±2dB measurement would require a bandwidth of over an octave. This is generally felt to be too wide, since it is much wider than the critical bandwidth of the ear, which is approximately 1/3 of an octave. In order to get better spectrum resolution, one that is consistent with the resolving power of the human ear, we need to supplement frequency averaging with another technique for reducing the measurement variability.

The second technique for reducing the statistical variance is to take several measurements over different spatial positions. This, too, will lower the variance if the measurements are sufficiently far apart in space. The rule of thumb is that each measurement should be at least $\lambda/4$ apart at the lowest frequency of interest. A good estimate of this would be about a foot apart. Roughly speaking, a measurement of the steady state sound field in a room will be about ±1 dB of the actual mean level if four 1/3 octave bandwidth measurements are made no closer together than about one foot. These dB levels must then be averaged together in dB (this is important since time averaging of a spatially moved microphone will not be correct). Narrower bandwidths will require more spatial points, and wider bandwidths will allow for fewer spatial points for a given resolution.

An example of these techniques is shown in Figure 4-3. This is a plot of the same room as shown in Figure 4-1, but with a frequency average of one-third octave at three points separated by one foot. This figure clearly shows that large variations in the response, even when frequency averaging is employed, can occur for spatially close points. Note that the 350 Hz band has more than 3 dB of variation in the response. The solid line is the spatial average of the three independent measurements, and it is expected to be accurate to approximately ±1 dB. The large variations between 100 and 200 Hz. are real and would need to be dealt with, but note how difficult this would be to correct. There are no other problems that stand out in this example. The low end looks quite reasonable in these curves, but at these low frequencies, one-third octave filters are not sharp enough to distinguish the modal variations in the response.

Another noteworthy feature is the much lower response at the higher frequencies. The response above 300 Hz is more than 3 dB lower than below 100 Hz. In the plots shown here, the frequency only goes up to 1 kHz. This is because of the huge calculation times required to go any higher. This is a

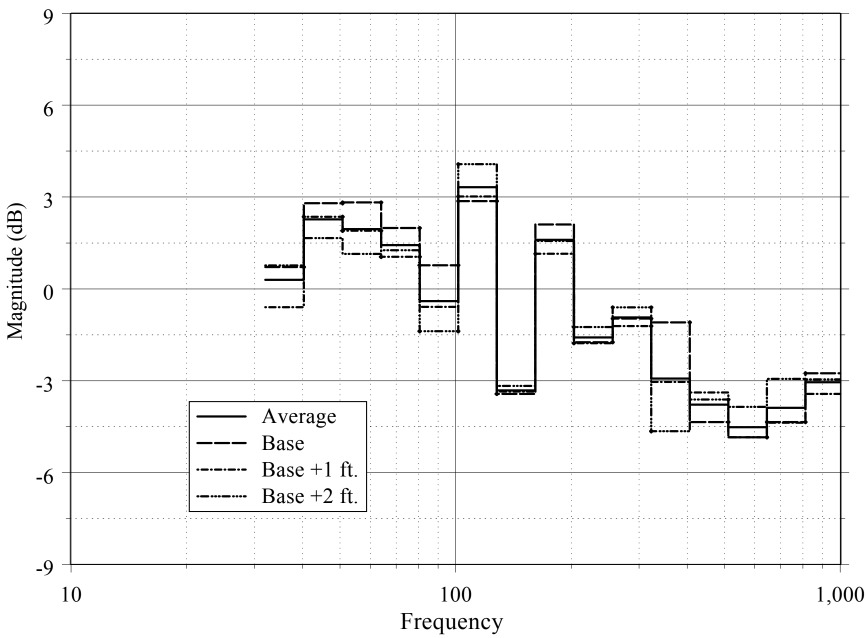

Figure 4-3.
One third octave frequency responses for a room at three nearby points

modal calculation in which there are 27,000 modes in the summation. In a real measurement one would normally take the frequency up to about 10kHz, which would be hundreds of thousands of modes.

I don't think that many people actually make acoustical measurements of this kind in their HTs, although it is getting easier and more common. It is important to note that while many installers do perform such tests, I have almost never seen one actually do a spatial measurement and average the results from several points (in dB!) when performing a sound system setup. Equalizing a sound system incorrectly can easily result in a poorer frequency response for the system than if no equalization had been performed. I recommend not equalizing a system in the statistical region unless it is going to be done correctly—using spatially averaged measurements and a knowledgeable practitioner. (However I will point out later that it is not the room that we really want to equalize anyway.)

4.1.d Low Frequencies and Damping

It is now time to consider the frequency range below f_S. To do this Figure 4-4 shows a new frequency response curve of the same room, but with about twice the absorption placed on the walls. This damping is ideal-

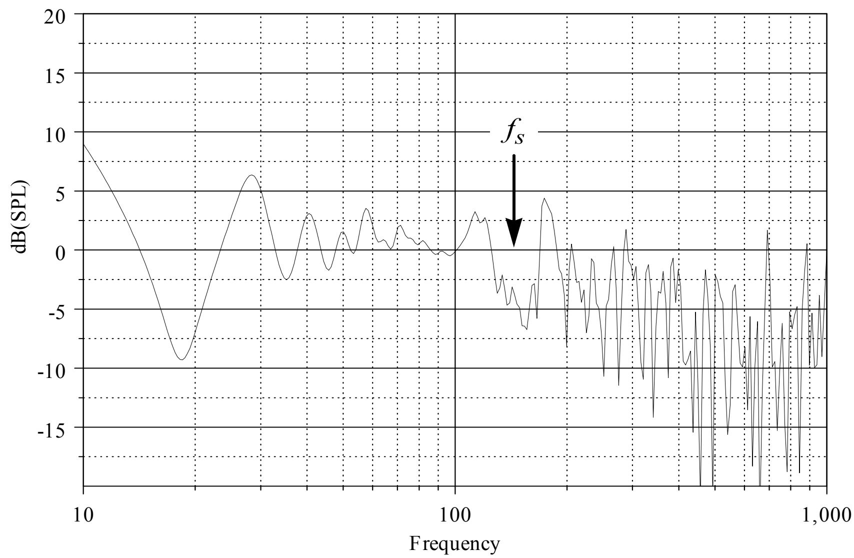

Figure 4-4. Frequency response in a room with high damping—in dB(SPL).

ized in that it is frequency independent, which is far from typical for the types of sound absorption in common use, and would be somewhat difficult to actually achieve in practice. Fortunately, as I will show, this type of frequency characteristic is not really what we will want anyway. It is used here for illustrative purposes only. There are several noteworthy features in this revised figure (compare to Figure 4-4 on page 61). First, the damping has a profound effect on smoothing out the lower frequency ($<f_S$) fluctuations, but it has a notably insignificant effect on the frequency response fluctuations above f_S. Above f_S, the mean level has been lowered, while below f_S there is no obvious reduction in the mean level at all. In the vicinity of f_S, a combination of effects is occurring which can be quite complicated to describe. Clearly, damping does a good job of smoothing out the response curve at low frequencies, but at higher frequencies all it does is lower the energy level.

Figure 4-5 (on pg. 62) shows the one-third octave response for the same room as shown in the previous figure as a direct comparison of the lightly damped, medium damped and heavily damped situations. These curves are all spatial averages of four locations. There are some important features to note in this figure. The lower energy level at the higher frequencies is quite apparent in the figures.

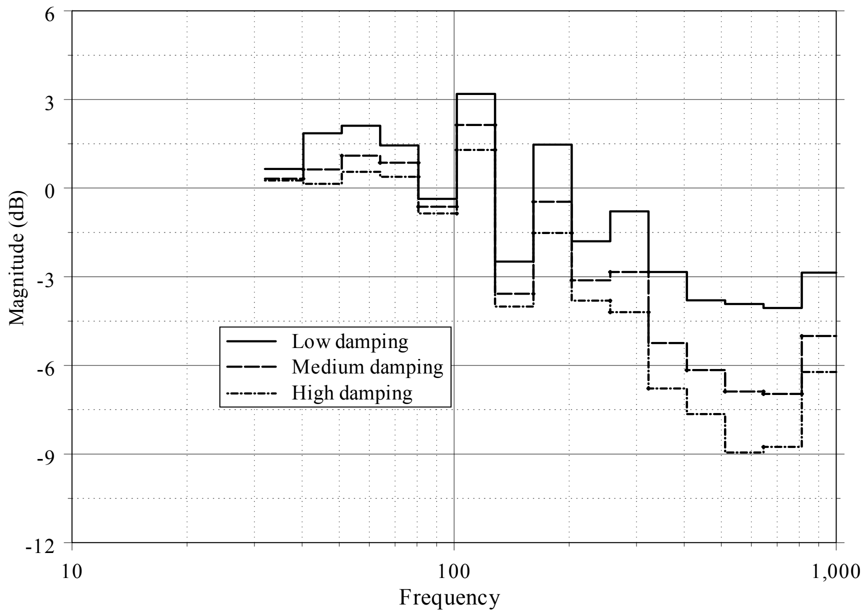

*Figure 4-5.
Room response with increasing amounts of damping*

At the lower frequencies, the increased absorption appears to affect the total energy level by only a small amount, 1–2 dB. Some smoothing is evident in the one-third octave plot, but the smoothing of the response fluctuations at these lower frequencies is much more evident in the narrow band curve shown in Figure 4-4. The damping has improved the 100–200Hz problem somewhat, but this region is still in need of some further attention.

The significant point here is that broad spectrum damping is mostly positive at the low frequencies—up to the point at which the statistical region begins to dominate—but at the higher frequencies no apparent benefit is evident from damping. There may be other reasons to use damping in a room, but from a steady state frequency response standpoint, the only benefits appear to be at low frequencies.

One other thing that I would like to point out before leaving the discussion of these figures is the strong inclination that one might have to call the peak in the response at 180 Hz and just below 700 Hz in both the narrow band room response curves (Figure 4-1 and Figure 4-4) a "mode" or a resonance—it is neither. It is simply a random event which has occurred in the frequency response curve at a particular frequency (unlikely to happen, but it is always possible—like tossing six heads in a row). Another indication of this fact is that adding damping does almost nothing to this sharp peak, a clear indication that it is not a resonance. There is also the fact that these peaks span several modes to tens of modes, not just one. In a spatially and frequency averaged curve, the lower peak is apparent, but the higher one is not. In fact, as the temperature in the room changes these peaks could actually disappear and reappear somewhere else.

It is important to understand the statistical nature of a sound field in a room. I have found from experience that this aspect of room acoustics is widely misunderstood. Explaining some of the implications of these principle is the reason for so many different curves and so many different ways of looking at the same room. More than anything else that I say in this text, the absorption characteristics that I recommend are the most controversial. This is because they are completely contrary to intuition, and quite contrary to what is typically done in large rooms (and most small ones) and what the marketplace recommends. I will show why intuition and convention in the design and treatment of a small room may lead to a sub-optimal listening environment.

4.2 The Direct and Reverberant Fields

A concept that also needs to be grasped is that of the direct field and the reverberant field. As a sensor is moved further and further away from a source in a free field its pressure response falls at –6 dB/DD (Double Distance). We have already used this fact on numerous occasions. This rule only holds true in a room to a certain distance from the source. Eventually the reverberation level of the room takes over and the steady-state pressure response in the room remains constant for greater distances. The transition is gradual. The –6dB/DD field is called the **direct field** and the constant field is called the **reverberant field**.

Direct field
that portion of the sound field that is dominate by the source level—no significant reverberation content.

Reverberant field
the region where the sound level is mostly reverberant energy.

I have shown a plot of this pressure response characteristic in Figure 4-6 for several qualitative descriptors of sound absorption. A rule of thumb that I use is represented by the straight dotted line. The point of intersection of this line with the pressure curves gives the point at which the direct field and reverb field energy level are about equal. Points left of this intersection indicate a dominant direct field and those to the right a dominant reverberant field. As a rule of thumb, the listener should not be in a dominate direct field location because at these locations there will not be a good mix of direct and reverberant energy required for spaciousness (see pg. 65). Note, that for lively rooms, this distance is at a reasonable seating distance from the sources, but for a dead room, one has to be so far away from the transducers as to be impractical or impossible.

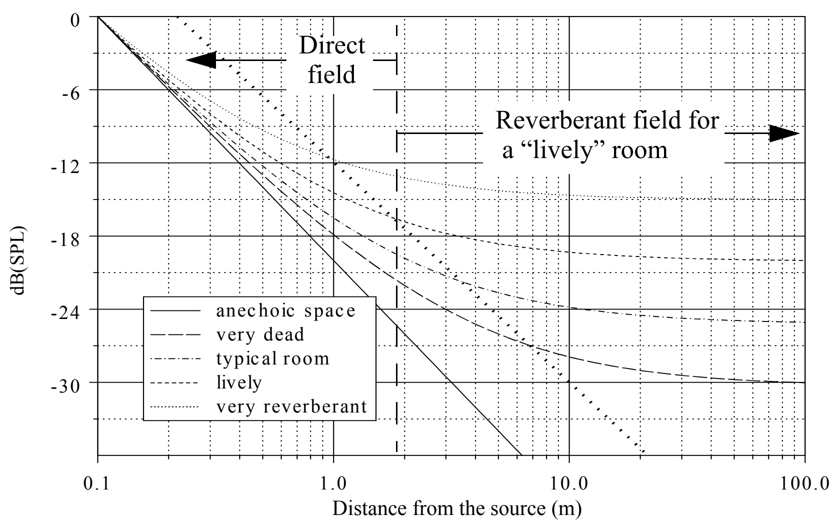

Figure 4-6. SPL versus distance from the source in a room, normalized to 10 cm.

The direct and reverberant fields are important concepts that will come up again in the context of noise control. It must be recognized that Figure 4-6 applies only to steady state sound. I should also point out that the statistical character of the sound field that I talked about in the previous section only applies to the reverberant field, it does not apply to the direct field. In a real room, there is always a direct field content until such time as the reverberant field has had sufficient time to build to steady state. The direct to reverb level and the time to reach this mixture are both critical aspects of the perceptual discussions that I will get into next.

4.3 Sound Perception in Rooms

Before I return to room acoustics design I need to define some requirements from the perceptual domain—drawing on some of the aspects of hearing that I touched on previously. To a large extent, the sound quality of a sound system in a room is characterized by three subjective aspects. They are its spaciousness, its localization capabilities and its timbre or coloration. The room can have a major, almost dominant, influence on these aspects of the sound system and expecting high quality playback without due attention to the room is naive. I will discuss these three criteria for a sound system by first defining what they are and how they are influenced by the room. Then, I will return to the acoustic problems that influence these criteria and address the room design to optimize them.

Spatiousness
the subjective feeling of being engulfed in sound.

4.3.a Spaciousness

Spaciousness is a purely subjective term that applies only to the room and not to the loudspeaker system. Spaciousness is the subjective feeling of being in an acoustically large space with a well distributed sound character. Precise definitions are not possible, but defining a room with a complete lack of spaciousness is—namely an anechoic chamber (see pg. 55). Spaciousness is created by a large number of laterally arriving sound waves which are delayed from the direct sound by more than 10-20ms. Only the reverberation field can possess this characteristic. The requirement for laterally arriving sound is apparent by considering the lateral placement of our ears and the enhanced ability of resolving sound arrivals in the lateral plan over other planes. Sound arrivals in the vertical plane are difficult for us to determine where they are coming from and, as such, tend to yield a more confused sound perception and are not considered significant or desirable to sound spatiousness.

In order to have the feeling of spaciousness, one must first be in a room location with a reasonably high reverberation level relative to the direct sound level. I showed this point in the previous section and how it varied with room absorption. As I showed in that section, in order to be at a reasonable seating location for good spaciousness, the room must not be heavily damped. A highly damped room causes an increased seating distance and an increased loudspeaker output requirement, resulting from the increased distance and the increased absorption. Spaciousness is just not possible in a heavily damped room. Once the correct distance and room absorption have been located, the time to build up the reverb field has to be considered. This time is both room and loudspeaker dependent. The more directional the loudspeaker the slower the reverberation field will build-up. Actually, calculating all of these requirements are complicated and so it is easier to deal with them in a more qualitative manner.

Loudspeaker Q
the ratio of axial response to the average off axis response.

An omni-directional loudspeaker sends sound in all directions, and, thus, there are many more early reflections from this kind of source than for the directional source. The directional sources takes a far longer time to build up the reverberation field. The directional speaker has a much higher ratio of axial response to power response than the omni-directional speaker. This ratio is sometimes called the **Q** of the polar response (the exact definition is somewhat more complex than this). An omni-directional loudspeaker has the lowest possible Q, namely one. The higher the Q, the more directional the loudspeaker and the slower the reverb field grows in time. This will be seen to be a major factor in the sound system design.

4.3.b Localization

Localization
the ability to sense the location of the virtual sound sources in a recording.

Localization refers to our ability to correctly locate the source of a sound in space. In our context here, it is our ability to visualize, in a hearing sense, the intended or virtual location of the sound emanating from the loudspeakers. This definition is predicated on the idea that the source material actually has a virtual source with an intended precise location. One cannot localize on a reproduced sound if the localization cues are not actually present in the source material. But many recordings do have intended virtual source locations, and the ability of the loudspeaker room combination to accurately place these sources is an important attribute of a quality loudspeaker—and a good room.

Localization is strongly affected by the loudspeaker-room interface situation. Localization in a large room is almost a trivial exercise since there are no early reflections to confound our localization capabilities. In this case the localization capabilities are simply those of the loudspeaker itself. Localization in an anechoic chamber would also be very good for the same reason - there are no very early refection to inhibit the ears localization function. It is easy to see that my three subjective criteria may be conflicting. In other words, good localization may come with degraded spaciousness and vice versa. A compromise has to be reached.

The principle problem with localization in small rooms are the early lateral reflections off of the nearest walls. These refections will be integrated (fused) with the direct sound and tend to confound the perception of the virtual sources location as well as cause coloration of the sound. Vertical reflections do not have a great deal of influence on localization since our localization capabilities are already poor in these directions, but they can have a strong impact on coloration as we will see in the next section.

4.3.c Timbre

Timbre
the tonal character of a sound.

Timbre refers to the tonal shift or coloration of a signal caused by the loudspeaker and room interface. This coloration can occur because of the early reflections, which create a comb filter effect. A comb filter results whenever two signals are added together after one of them has been delayed in time. It is sometimes called multi-path. An example of a comb filter is shown in Figure 4-7 on page 68, where the second signal is delayed from the first by about 1 ms.

There are two critical aspects of a comb filter for small time delays. The first is that the notches are spaced wide enough that they fall into alternate critical bands, thereby being perceivable as frequency effects. The second is that the small time delays puts the second signal within the ears integration time, thus fusing the two signals together. The ear is actually receiving the filtered response as shown in the figure. When the time delay is extended, the notches become so dense that they are basically not resolvable by the ear, and, second, they are not as readily fused into the perception as the shorter delays. The net result is that the ear is highly sensitive to short time delays, and this sensitivity drops with longer delays.

Timbre is much like localization in that it is strongly affected by the early refections. In one sense, timbre and localization degradation due to time delayed signals are the same thing, but they can also be different. A

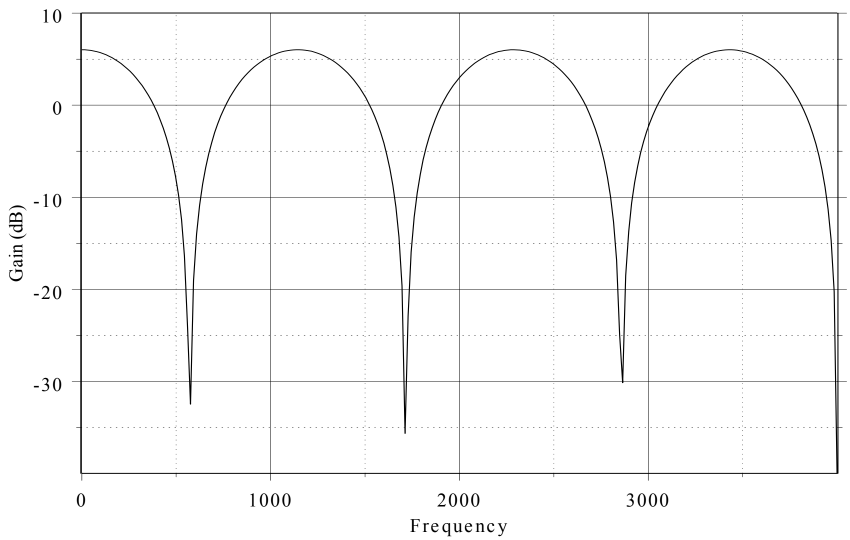

Figure 4-7. Comb filter resulting from the combination of two signals delayed from each other.

vertical reflection causes strong coloration but is not a significant localization problem. Lateral refections cause both. Timbre problems can also come from the loudspeaker itself in the form of non-flat power response (where the direct sound and the reverberant sound do not have the same timbre) or strong resonances, which don't tend to affect the localization as much, but do cause significant coloration problems.

It should be pretty clear by now what we want and what we don't want in the way of small room acoustics. Below is a summary:

- First, we want a large amount of sound absorption at low frequencies to help smooth out frequency and spatial response irregularities. This absorption helps what is called the modal overlap or the extent to which the modes interact. Absorption helps to "mix up" the low frequency modes.
- We want as little absorption at higher frequencies as we can reasonably get away with. That's because, in general, we do not see any positive effect on the frequency response from generic absorption, and we know that it will degrade the perceived spaciousness of the room by removing the desirable multiple lateral reflections.
- We want to try and eliminate early reflections or at least minimize them as much as possible. I have shown how the smaller the reflec-

tion delay, the more detrimental it tends to be and extending the reflection delay time while reducing its level is highly desirable. I have not yet discussed how to achieve this, but there are options.

- We will want to be careful not to go overboard reducing the refections because that will simply lead to a dead and lifeless room.
- An early reflection arriving at an alternate ear is not as bad as an early refection arriving at the same ear. The former case has a binaural advantage in the brain's signal processing that the later case does not. This fact makes most reflections in the vertical plane undesirable, but again we have to balance this requirement against the desire for little high frequency absorption.

Lets now return to our discussion of the absorption in a small room. I have shown that it is desirable to have large low frequency absorption with little high frequency absorption, where there may be a few exceptions used to control specific early refections. In a practical sense, there is a real problem with this requirement. Virtually all acoustical treatments for rooms have large high frequency absorption dropping to almost nothing at low frequencies, which is exactly the opposite of what we want. Clearly, dealing with the absorption aspects of a room by the use of standard materials is not recommended. The use of sound absorption in a small room must be dealt with extremely carefully. It has been my experience that it is almost impossible to make a small room too live at high frequencies. Most typical room construction materials and furniture have significant levels of absorption at high frequencies. Obtaining the right amount of absorption across the frequency band requires different construction techniques and room interior treatments.

I will return to the construction details in a later chapter, however, there are some specific topics that are more relevant here. How absorption actually works is an important issue in our current discussion. There are two principle mechanisms for sound absorption.

The first is to use a porous material such that the sound wave can penetrate it, and, in doing so, the air moving in and out of this porous medium dissipates energy through friction. This mechanism is by far the most common, and there are some specific features to this kind of absorption. First, it becomes increasingly less effective as the wavelength of sound exceeds the thickness of the material. Thin materials will have no low frequency absorption. The second is that since the porous material works on the acoustic particle velocity, the effectiveness of the material is reduced when it is placed at locations of low particle velocity—places like walls where the velocity must go to zero. A piece of sound absorbing material placed on a wall is 1) not very effective and 2) increases in effectiveness as the frequency increases. This most common of all sound treatments is exactly the wrong thing to do.

The second major source of sound absorption is through the actual motion of the room structure—the walls themselves. Of course, if these walls are perfectly rigid—like poured concrete—then this mode of absorption is negligible. But, for a common frame and dry wall construction, wall motion can be quite substantial. Since the wall has mass, its motion will continue to fall as the frequency goes up—that is, unless it has a resilient support structure. All walls must be supported in some way. When the support is resilient (and all supports are to a certain extent, except for maybe a concrete backing), then there will be a resonance frequency and the motion of the wall will fall both above and below this resonance. A wall would typically resonate somewhere below 100 Hz—depending on drywall thickness and the method of mounting. When the wall does move, it dissipates energy through friction. (All absorption is friction of some sort.) The main difference with this type of absorption is that it decreases with frequency rather than increase as the porous material method does. This would seem to be the ideal mode of absorption for a small room and indeed it is. In fact, if done properly tremendous absorption can be achieved at low frequencies with almost no high frequency absorption.

Another concept in sound absorption that comes into play in most HTs that I have done has to do with sound absorption on opposing walls. In my book *Audio Transducers*, I show how, at low frequencies, sound absorption works the same whether it is on one wall of an opposing pair or it is on both of them. By this I mean that the sound absorption is the same whether it is split between two opposing walls or all of it is placed on one wall. This is a good thing to know because it means that if we need to add low frequency absorption to a room, we need only do it on one wall of each of the three opposing pairs. I will show how this is a major advantage when locating a HT in a home.

In small rooms, the obvious preferred mode of sound absorption is to have the walls constructed in such a way that they deliberately move and absorb sound. This technique would never be used in an large auditorium because in those venues we are looking for primarily higher frequency absorption and hanging type materials, such as curtains or drapes are a good choice. In a small HT, the walls should be bare and hard but mounted so that they flex at low frequencies.

4.4 Summary

A complex chapter like this deserves a summary of its main points.

- The first is that the sound field in a room is a random quantity and one must do some form of averaging of samples (frequency responses) to get a valid estimate of the true frequency response.
- The modal region of a room, where there are true resonances, is limited to a relatively low frequency region never more than a few hundred Hertz.
- Damping in a room is effective at smoothing out the modal region but is detrimental to the steady state response at higher frequencies.
- Sound quality is strongly affected by early refections and the level of the reverberant field, and, in a small room, these two requirements can be in conflict.
- The loudspeakers directivity will strongly interact with early refections and the onset of the reverberant field.
- Most commercial sound absorbing materials do not have the correct frequency characteristics for a small room—they work at high frequencies but not low frequencies.
- Wall internal damping is the ideal way to dampen a small room because it can have good low frequency absorption with little high frequency absorption.