Basso Profundo

By [Martin Colloms](#) • Posted: Dec 29, 1991

Bass constitutes one of the least understood aspects of sound reproduction. Opinions vary greatly on matters of bass quality, quantity, and perceived frequency range or response. Moreover, the bass region is subject to the most unwanted variation in practical situations due to the great influence listening-room acoustics have on loudspeaker performance. Every room has its different bass characteristic, and changes in the position of speakers or listener also constitute major variables at low frequencies.

What constitutes good bass?
Limits: For this discussion, the bass range has been taken as 20Hz to 160Hz---no less than three octaves. (Hz = Hertz, or cycles per second.) We know that bass sounds can occur low enough in frequency---in the 20-30Hz region---to be felt as well as heard. The sensation of pitch is lost below 20Hz, and we hear instead the individual bumps of the acoustic pressure waves. The vast proportion of recorded music has little content below 40Hz, but when music does venture down into the lowest octave---and I repeat the term "octave"---big audio systems can reproduce subjectively exciting things. The program acquires scale and weight when this range is properly reproduced. A large orchestral bass drum can sound strongly in the 35Hz area, and while the bottom E on a conventional double bass or electric bass guitar is customarily 42Hz, synthesizers and a large pipe organ can play much lower, to 20Hz at least. Fig.1 shows the frequency ranges covered by common instrumental sounds.

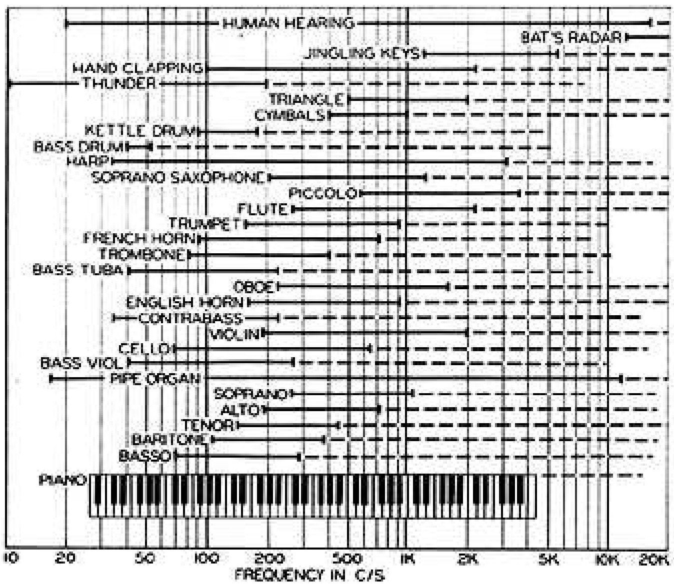

Fig.1 Frequency range required for realistic reproduction of various sounds (after *The Audio Encyclopedia*)

Slam: Several subjective characterizations are relevant in the bass. "Slam" is an interesting one, for me conveying an awareness of a massively realistic bass percussive sound---one which is deep, fast, and yet has a close association with the upper harmonic information that conveys the instant of impact. "Loose" bass indicates a softer, slower bass separated from the impact. In comparison with real life, can the audio system effectively reproduce the heavy slam of a big door? At least one system can: the WATT/Puppy, in which bass speed was a crucial design factor. Accurate presentation of time and rhythm information is this system's forte.

Timing: "Timing" is a term gaining increasing importance. "Good timing" generally indicates that temporal aspects of the music---tempo and rhythm---sound realistic.

Rhythm: Fundamental to rhythm is the perception that the rhythmic aspects of the music, particularly in rock and jazz, are presented in unison, coherently locked together over the entire frequency range. For example, a good pattern on a snare drum must be allied to pedal bass drum *on* the beat; likewise, the complementing sounds from the cymbals must tie in with the other two. When timing is right, the whole sound seems to slide into temporal focus. One is aware of the rhythmic flow, the ebb and surge of the syncopation, the expressive playing of great performers. For many listeners, this is just as important as the optimum reproduction of the classic aspects of stereo staging, transparency, neutrality, and detail.

When all is right, the reproduction is imbued with an appropriately realistic sense of pace, the music driving forward with life and energy, confidently drawing in the increasingly involved listener.

Slow, soft, loose, overhung bass is a killer in this respect. Designers who specialize in getting good rhythm have learned that if a given system cannot reproduce bass well, it's better to leave well enough alone and not attempt the impossible. Stiffening a small speaker's suspension prevents the driver cone from moving too much. In direct contradiction of the classical theory, controlled short throw (and, by implication, a restricted bass response to, say, above 55Hz) has the following benefits: The upper-range bass can be designed for better damping since the whole system can be adjusted for a higher conversion efficiency.

Reduced cone excursion, especially for drivers working over the bass and midrange, has the benefit of reducing a host of modulation effects both dynamic and magnetic. Higher efficiency promotes lower voice-coil temperature and improves subjective dynamic accuracy. Distortion is also improved since it is strongly proportional to input power, this much reduced with higher system efficiency. Abandoning the lower bass register allows the smaller system to use a lighter cone with a superior transient response in the midrange, again contributing to system "speed."

Striking the balance between classical virtues and timing

How do you determine the optimum balance between life, pace, and speed, and bass extension, low coloration, and transparency? The last three are classical virtues of a high-quality audio system, yet the first three are also vital if those intellectual qualities are to remain satisfactory. A good novel is not just fine prose---it must also have a gripping story if it is to entertain. Indeed, a great story can still be involving even if the writing is not so good.

Why that stress on the word "octave" in the introduction?

Audio engineers concentrate much of their attention on the midrange, officially centered on 1kHz but in practice encompassing the 200Hz-2kHz decade, about four octaves in all. A great deal certainly happens in this range in musical terms; the majority of sounds, voices, and musical instruments play their tunes here. However, the bass is often neglected by comparison, possibly because the number of Hertz is less. It's easy to imagine a full musical scale over the 400Hz-800Hz range with 400Hz to play with. The 30-60Hz bass range has just as many tones and semitones, etc., but makes do with only a 30Hz interval. We need to view low-frequency responses with greater precision, and try to discount the idea that only higher numbers of Hz matter.

A quick look at the treble really shows this up. For most purposes, the 5kHz range from 15kHz-20kHz may be discarded, and very often is, by a combination of source spectrum, microphone, filters, and loudspeakers, many of which are quite ineffective above 12kHz. Compare that 5kHz half-octave bandwidth of upper harmonics and noise with just 50Hz worth in the bass range below 100Hz. This latter span is a whole octave of fundamental bass playing, in most works representing the foundation for musical composition. Rock could hardly exist without it. Every 10Hz of good, even, controlled bass extension matters in the design and specification of loudspeakers, and makes their design and evaluation such a subjective matter.

Perception of bass loudness

To understand why reproduced sounds are not always heard as one might expect in the bass, a closer look must be taken at the curves for perceived loudness. These are often used in an oversimplified way (see fig.2, which shows the free-field equal-loudness contours developed by Robinson and Dadson). In general, a discussion about audible loudness steps concentrates on the midrange, a region of naturally high sensitivity. This can be misleading where the bass region is concerned. There is a wealth of fascinating detail in these curves, and it is important to arrive at a better understanding of their implications. This part of the discussion was promoted by a casual aside made by Floyd Toole, of Canada's National Research Council, during a recent lecture on room acoustics at a meeting of the London AES: "Of course, we are more sensitive to small variations in bass level than we are in the midrange." In a related discussion, he also noted that a little bass appears to go a long way subjectively, or words to that effect. These two observations have serious implications.

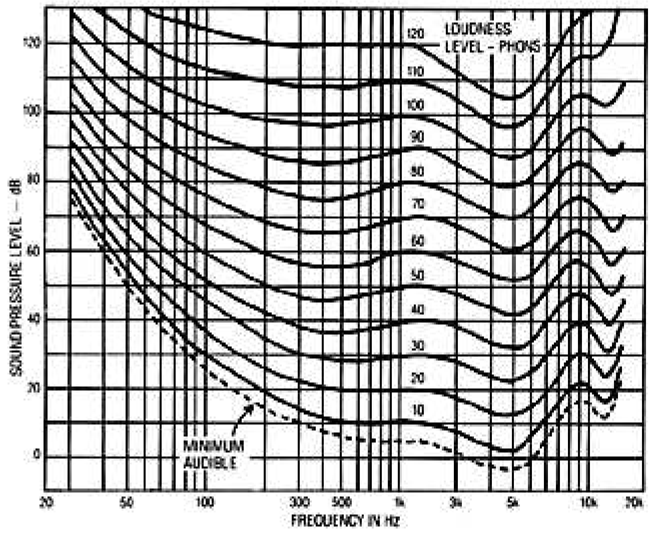

Fig.2 Robinson and Dadson free-field equal-loudness contours (after *The Audio Encyclopedia*). Low-frequency sounds have to be played at a much higher SPL to sound as loud as mid-frequency ones.

First, it should be pointed out that our absolute sensitivity to bass sounds is much poorer than in the midrange. Evolution has provided us with a tailored aural sensitivity tuned to the real world: twigs breaking, rustles in the grass, the calls of animals and birds. The threshold of audibility is defined as a sound pressure of 0.0002dyne/cm at 1kHz, and "0dB" is assigned to this level. Between 3kHz and 4kHz, our hearing is still more acute; many younger listeners can detect sound levels down to -10dB in this region. Moving into the treble range, by 10kHz some loss in sensitivity is normal (though such loss increases with age). Nevertheless, levels down to 12 or 15dB above the threshold at 1kHz are still audible, if very quiet. However, if we look at the low-frequency region, the picture is very different. Consider that dotted threshold sensitivity line.

Even by 150Hz the typical sensitivity has fallen below that for 10kHz. To be audible, a 75Hz tone in the midbass must be 30dB louder than the 1kHz, 0dB threshold, while 50Hz requires 40dB, 100 times the pressure. The required level for audibility at 25Hz is over 60dB, one thousand times, although it is worth noting that the test results are fundamentally based on headphone listening and that the whole-body excitation which results from room-propagated bass, which can tend to enhance bass sensitivity, is absent.

As a direct consequence of these curves, there are also changes in tonal quality which occur with music according to the volume level at which it is reproduced. The subjective loudness curves represent a complex dynamic variable in which an envelope representing a musical section must exist. That envelope bounds the frequency response and loudness variation of that passage or whole work, and it must be located somewhere in the hearing response curves. As Peter Walker has concisely stated, *"there is only one correct volume level for any particular piece of music."* Taking into account listening-room acoustics, the original recording technique, etc., in theory there is just one volume level which places the dynamic music envelope correctly in the characteristic curves. A particular instrument was played at a particular loudness, and a matching tonality should be reproduced at a matched level relative to the listener and his expected location. Only then will the instrument sound natural, and the normal dynamics of musical expression be reproduced in the expected range.

Thus the perfectionist user of a volume control should learn to reproduce the correct, natural playback level for each and every recording.

Setting any other volume level, such as for the comfort of one's family or neighbors, may be judged a distortion of the truth. One is making a convenience of the reproducing system and partially devaluing the musical message. Conversely, one could argue that in a concert hall one can personally arrange for a different performed loudness and resulting tonal balance by selecting seats nearer or farther away from the performers. Here, as with reproduced volume, taste and preference also play their parts. In recording practice, the taste and judgment of the balance engineer is a dominant influence where the distance between the microphone and the performers is the major variable.

It is an accepted part of our experience that sound sources not only get quieter the farther away they are, but also that distance lends an appropriate change in tonal quality. Only a small part---a degree of treble softening---is due to specific high-frequency losses incurred over the intervening air path. The bulk of the change in perceived frequency response is due to one's ears, the placement of the sound envelope further down in the set of characteristic curves. Take, as a reference, an 80dB level; this is typical of loud nearby speech. Now consider the subjective frequency-response changes in the frequency range below 800Hz, the latter approaching the highest practical compass of a soprano. Comparing the 80dB level at 1kHz, the sensitivity at 180Hz is also on the line. However, when the overall sound level is set 20dB lower to 60dB, this 180Hz frequency is now heard 2-3dB more quietly than it should be, while at a 40dB intensity the relative loss has increased to 4dB. By a whisper-quiet 20dB, the relative loss has increased to 8dB, additionally accompanied by several dB of associated loss in the lower-mid frequency range centered on 500Hz.

As a result, a more distant, quiet source sounds "thinner" and "lighter" in tonal quality due to these clearly audible changes in subjective frequency response. For most of us, it comes so naturally that we are not really aware of it.

Thus one classic interpretation of the action of a volume control is the control of distance. Reducing the reproducing level alters the loudness and the simultaneously perceived tonal balance, giving the imprecision that the musicians are farther away. While this view is valid for classical program played at a natural level, the argument is weakened when rock music is involved. Very few, if any, of the sounds in modern mixes have a truly natural basis, hence the foundation for the aural recognition of tonal-balance accuracy is lost. With rock the case would seem to be the louder the better, provided that you and the audio system can cope. Nevertheless, this question of natural loudness is highly relevant to music played on acoustic instruments, including the human voice.

The timbre of an instrument is a highly variable quality, and such variation is skillfully exploited by composers and musicians for expressive purposes. Setting aside matters of musical technique---the position of a bow along the length of a violin string, or the particular selection of a reed for woodwind---there are factors which will affect sound quality and timbre, according to the loudness at which they are played. Timbre will change according to the physical energy put into a performance. This is a natural part of the dynamics of an instrumental performance and must not be masked or exaggerated by a hi-fi system. If this happens, the natural musical dynamics will be falsified. Several ailments result, including the sin of "compression." In the real world there is a general tendency for audio equipment to impair natural musical dynamics. It is quite natural for the timbre or tonal quality of an instrument to become harder, fiercer, brighter, and sharper when it is played more strongly, with greater energy, and at a greater volume.

Knowing that the perceived frequency response for the ear varies with loudness, it can be seen that our convenient logarithmic dB scale for loudness is a serious oversimplification. Looking at the curves, one can pick one convenient frequency, 1kHz, and see that the dB steps in phons or audible loudness pretty well match the log scale for sound pressure in this frequency region. However, they don't fit so well at 500Hz. For example, the aural response shape at 120dB is not repeated accurately down the dynamic range. A similar fault is present at 4kHz, where the response dip is deeper at high than at low loudness levels.

Bass at last
At last we get to the bass region, and here the characteristic is anything but logarithmic. Both spacing and the shape of the curves vary continuously with loudness. Moreover, we must remember the high audibility threshold, below which bass cannot be heard at all. Put another way, we all suffer from an increasing rate of bass rolloff with decreasing loudness. This fact has practical consequences. There's little point in producing a miniature loudspeaker of modest loudness capability which boasts an extended bass performance if the little darling can't play loudly enough for its low notes to be audible. Instead the designer should exploit the relationship between the bass extension and sensitivity, curtailing the former until the available loudness rises to a point where that bass which can be reproduced is usefully audible.

Simplifying the matter and assuming that typical room gain of 6dB by 40Hz, at the listening position a speaker will need to be capable of delivering 100dB if full-weight, 40Hz bass is to be audible with normal orchestration. At 40Hz the sensitivity threshold is 54dB, the sensitivity loss at 100dB being around 6dB. This leaves a dynamic range of 46dB, or the bass extreme compared with the 100dB available (in theory) in the midrange. At sound pressures above 90dB (1kHz, 0dB reference) the bass response becomes more logarithmic. Below 70dB, the bass lines in the characteristic responses begin to bunch up. This bunching reflects the fact that we hear changes in bass level more acutely than in the midrange.

This increased sensitivity for change could be interpreted as a compensation of the reduced dynamic range, but it is also the basis for that earlier comment that "a little bass goes a long way." Once we get into the working bass area, the ear is more sensitive to variations in uniformity and absolute balance. This is why there are more complaints about bass---whether too much or too little---than in other parts of the frequency range.

In a large hall, bass may propagate freely down to the lowest practicable frequencies; there is an open, easy, flowing quality to concert-hall bass. By contrast, listening-room bass is often lumpy, boomy, and oppressive when in excess, while in unfortunate locations it can seem infuriatingly absent. The dimensions of normal rooms are comparable with the actual propagated wavelength of the bass sounds we are attempting to reproduce. Those infamous standing-wave modes are the problem, where the distribution of bass energy varies strongly with speaker and listener position as well as with frequency. Major variables include room size and shape, and the stiffness of its boundaries. Bass is subjectively much more powerful in a brick-and-concrete building than in a frame house with stud-and-plaster walls and suspended timber floors. In many traditional American frame houses the bass energy simply leaks away through the acoustically soft structure.

Optimized speaker location

While some speakers are designed specifically to be mounted close to the wall, the majority are expected to be used in free space. What exactly does this mean? As regards acoustic theory, free space means just that---the great blue yonder---with the speakers located well off the ground, say 100' for starters. In the context of a speaker in a room, however, free space simply means moving the speaker clear of the back and side walls and mounting it on an open-frame stand appropriate for the type and at a worthwhile height. With a listening position somewhere in the last third of the room, even up to the back wall, the speakers on the other hand would be placed in the region of 1/5 to 1/3 of the way out from their nearest boundaries.

In a room 13' wide, 17' long, and 9 ½' high with an average low-frequency loss, this means the speakers will sound tolerably well-balanced if placed with their radiating centers about 2' from the floor and 3 ½' from the side walls, plus 2 ¾' from the back. Note that these three dimensions are unequal, with the deliberate intention of avoiding a coincidence in spacing to these three local boundaries. Interferences from the energy reflected by the local boundaries is thus spread over a range of frequencies, reducing their effects. While interference produces sharp notches in the frequency response, the general action of the local boundaries is to act as reflectors, progressively boosting the bass level. This boost is real enough, typically adding 6dB in the 30-40Hz range. This is highly convenient: If our compact speaker is well-tuned in the bass and critically damped for optimum transient performance, it can be designed to be typically -6dB in absolute response at the low-frequency limit. This is why good small speakers, when properly aligned and placed, can produce reasonable bass down to 35Hz in many rooms.

Room boundaries

In order to provide an agreed basis for the mathematical analysis of the low-frequency response of loudspeakers, academic acousticians made the assumption that loudspeaker theory is based on "2Pi" environments. The "Pi" nomenclature refers to the solid angle, in radians, into which the speaker fires: 4Pi is thus free space, conferring truly spherical radiation at low frequencies, while 2Pi represents a hemisphere where the speakers are effectively flush-mounted in an infinitely large wall. However, a room is rather more contained than this, and at frequencies below

500Hz the radiated sound wavelengths are sufficiently large to suffer both cancellation and boosting, or reflection, from the local boundaries.

For a "free-space," stand-mounted speaker system, the first cancellation is at approximately 160Hz and is due to the floor. The back wall comes into play next, followed by the side walls. In-phase reflection results in a theoretical boost of 3dB per boundary; with idealized non-coincident spacings and good room proportions, the maximum lift could be 2.5dB at 80Hz, 6dB by 50Hz, and 8dB by 35Hz. This underlies the uneven but progressive room gain present at lower and lower frequencies. In normal rooms, it explains why a small speaker with a free-space response 6dB down at 50Hz under test chamber conditions can still produce some audible 40Hz in the listening room. It also explains the oft-noted difference in bass quality between traditional acoustic-suspension and bass-reflex systems. The former have been described as providing more even, more extended, and less colored bass than the latter.

Traditionally, reflex systems have been described as "boomy," yet are often found to provide less power in the low bass than the specifications suggest. Those specifications were based on test-chamber measurements and a rated -3dB rolloff in the bass. However, the important factor turns out to be what happens *below* rolloff. The aural sensitivity to changes at low frequencies is great, and we have room gain adding to the available response, helping to counteract the rolloff. A well-damped sealed-box system can have a desirably slow rolloff rate significantly complemented by room gain, thus extending the overall useful response. Below box resonance, the output from a bass-reflex system usually falls rapidly, too quickly for the room boundaries to help out; no low bass is heard. In addition, the reflex system is likely to have a sharper, squarer response "cover" at rolloff; mild room gain at this frequency can easily turn the corner into an audible lump---the notorious "boom."

Variation due to speaker type

There is insufficient space here to cover various speaker types, though the open-panel dipole is worth some attention. A box speaker's lower range increases in level as it nears the floor, wall, or corner. Perversely, for a panel speaker---Magneplanar, Apogee, and particularly the super-light diaphragmed electrostatic type such as the Quad---the effective low-range output is decreased. When very close to the wall, the output from a Quad approaches zero as the listening back wave is in opposition, out of phase with the moving element and almost canceling out the audio output. Moreover, the cancellation is anything but uniform, as the cancellation zeros appear at multiples of the audio wavelength. This "comb" of notches presents quite serious coloration and explains why the positioning of panel speakers in rooms is highly critical.

Generalizations are not possible here. Full-range electrostatics perform best well away from the back wall. For example, the Quad ESL-63 can produce serious bass in large spaces, but becomes lightweight when backed up in a small room. Conversely, a full-range Apogee requires either a smaller room where a tendency to bass excess can be judiciously tamed by exploiting some back-wall cancellation, or a very big space where little or no room lift is present in the main low-frequency range.

Active control

Devices are available which can be placed in regions of greatest low-frequency standing-wave energy, the Phantom Acoustics Shadow reviewed by RH in Vol.12 No.12 being the only example commercially available in the US. Set correctly, these monitor the incoming waves and generate anti-phase energy, sapping the power of the standing wave. If used with care, this expensive technology can be quite effective.

Summary

Opinions vary greatly concerning reproduced bass sound. Many factors play their parts, from such acoustic fundamentals as the standing wave or resonant behavior of these long audio wavelengths in small listening rooms, to the influence of loudspeaker type (open-panel or box designs). Our hearing characteristics also play a part, with a combination of poor absolute sensitivity in the bass and greater sensitivity to changes in level in the audible range.

Subjective characterization of bass quality is important, determined both by the final frequency response perceived by the listener and its qualities of time coherence with the remainder of the frequency range. A rhythmically involving if restricted bass may prove of higher quality in terms of enjoyment than a more extended bass register lacking pace and speed. Sheer bass quality, in level or extension, is no yardstick by which to measure musical quality. The negative influence of the normal technical presentations intended to depict bass sound quality needs to be appreciated. In the usual frequency-response graph the bass region is not accorded its rightful visual weight. In the light of the psychoacoustic responses there is a good case for weighting the bass region by an expansion of, say, 1.5x in amplitude and 2x in frequency on the graph, to allow the observer to make more accurate assessments of bass performance.

At very low frequencies, every Hz counts!