Chapter 14. Room Acoustics

Background

Thus far, many conditions which affect the performance of a loudspeaker system have been examined. The drivers and the enclosure also interact with each other and the surroundings. This can affect the frequency response of the designed system. Listening rooms display vastly characteristics to anechoic rooms hence there is a need to evaluate the influence of the room boundaries and modes on the frequency response. The 'Room Acoustics' option with its three screens helps to understand these issues. Assumptions are as follows [5]: (1) The loudspeaker behaves like a monopole sound source, (2) The average absorption of the room surfaces is more than 3% and (3) The three closest boundaries are almost perfectly rigid reflectors. Most normal rooms will easily fit into these descriptions. The positions in which the loudspeaker systems are placed within the room are often dictated by the convenient spaces which happen to be available rather then by any scientific rules. However, if the best possible quality of sound reproduction is desired from the system, it needs to be positioned subject to a better understanding of the influencing factors. If a loudspeaker is moved about in the corner of the listening room, the user will notice, that the increase in the power output at lower frequencies can be as high as 9 db. This increase in the power output of the source can be explained with the help of the radiation resistance.

The efficiency of a typical moving-coil direct radiator loudspeaker system is in the order of 1-5%. This poor efficiency is due to the relatively small mechanical load which the air presents to the diaphragm and it's associated enclosure. Hence, most of the input power is dissipated in energy losses by heating the voice coil. A small percentage only is consumed by working against the resistive part of the air load (radiation resistance) to generate acoustic radiation. When the loudspeaker system is placed near one or more room boundaries, a part of the sound wave radiated by the loudspeaker diaphragm is reflected by boundaries back on to the diaphragm. If the wavelength of the sound radiation is much larger then the distance between the loudspeaker diaphragm and the boundary, then the combination of the outgoing and the reflected waves results in an increase in the radiation resistance.

Because the radiation resistance is only a very small part of the total mechanical impedance of the system, the amplitude of the diaphragm velocity will be virtually unchanged. This constant diaphragm velocity and the arrival of the reflected sound waves in-phase with the radiated sound increases the radiation resistance and then the power output of the system particularly in corner position. As the frequency is increased, radiation resistance falls until the reflected sound waves arrive in exact opposite phase to the radiated sound wave. This creates a severe dip in the power output of the system, as the air load becomes very small.

Enclosure Placement

The "**Room Image Method**" - enclosure positioning, function is incorporated under the **"Room/Car Acoustics**" menu and invoked from the main menu of this module (see Fig 14.1).

The first button - "Clear" is used to clear the plotting area. The next seven buttons provide plots for the indicated enclosure type. "Your Woofer" button recalls/generates plot for the woofer which has been created as the amplitude response (Transfer Function) with the "Editor" tool and "Your System" curve must have been created by the user as a summed response of the configured system. Box distance from the three boundaries (corner of the room) is selected using three scroll bars, one for each direction: X, Y and Z. Every time one of these bars is activated, the corresponding position of the enclosure is changed by a small increment. The minimum distance allowed is 5 cm, and the maximum distance allowed is displayed on the graph at the end of each X,Y,Z coordinate. The graphs are labeled automatically. You can display the woofer frequency response in various modes (enclosures) or your system frequency response. Figure14.1 shows an example of enclosure placement and the resulting responses. If a loudspeaker measured in a true free field, with no reflecting surfaces, such as a very good anechoic chamber, or outdoors far away from any large object, including the ground, the sound radiates into a full sphere solid angle, identified as 4π steradians. Placing the loudspeaker on a floor reduces the solid angle by half, and the sound pressure at low frequencies will be increased by approximately 6 dB because sound that would have propagated away will be reflected back. Leaving it on the floor and backing it up against a wall reduces the solid angle by two factors of two, to π steradians and the sound pressure is elevated by about 12 dB. Sliding the loudspeaker into a corner reduces the solid angle into which the sound radiates by a further factor of two, to $\pi/2$ steradians, and the pressure goes up by another 6 dB for a total low-frequency gain of about 18 dB! With respect to amplifier power and stress and strain on the loudspeaker, these acoustic gains are absolutely free, so wise people take advantage of them. An unfortunate side effect is that some irregularity is introduced at upper bass and lower mid frequencies.

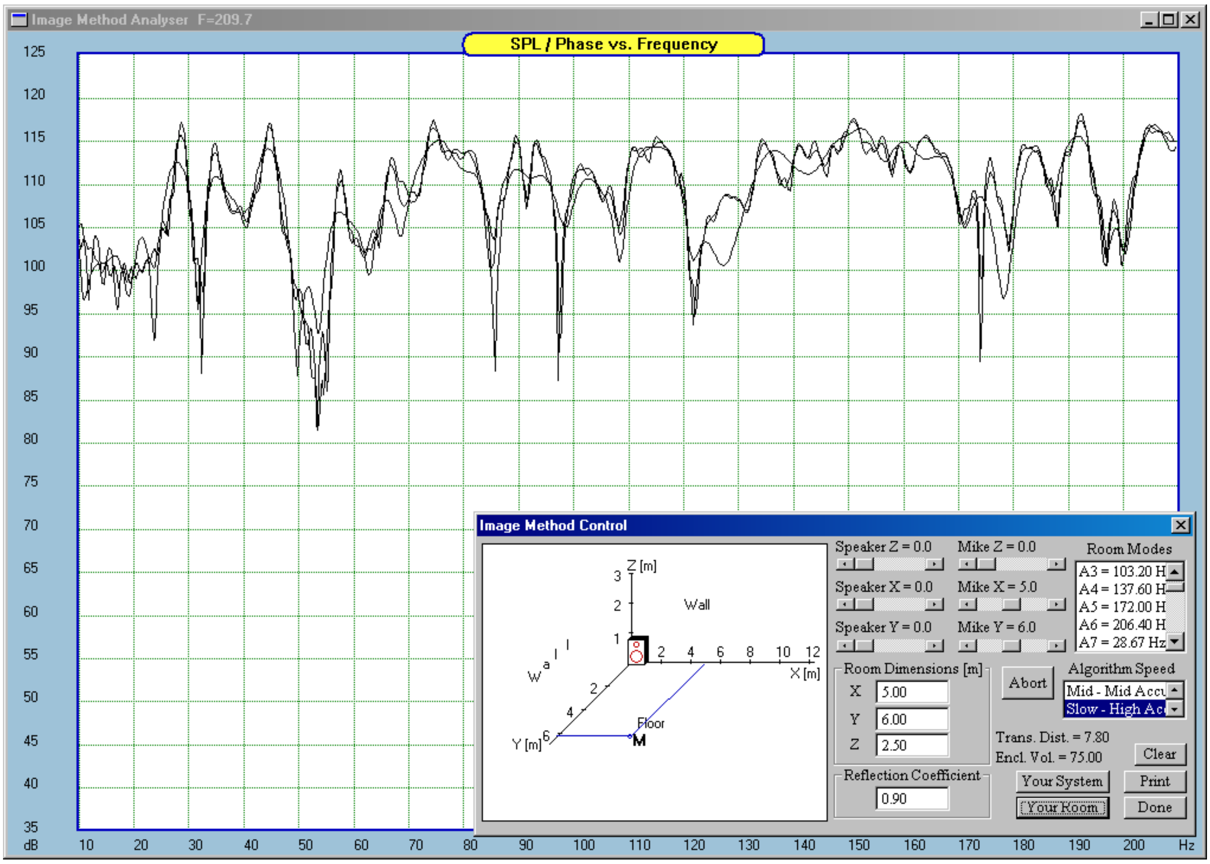

Fig 14.1 Image Method Screen

Image Method

Behavior of the sound waves in a small enclosure is a quite complex problem. However, the wave property of the sound energy makes it possible to introduce significantly simplified model of the problem, restricted to low frequencies, where only the room dimensions are of significance. In a rectangular room we are going to consider, the sound waves will be reflected and re-reflected between all 6 surfaces. When the reflections occur between opposite surfaces, the standing wave pattern generated this way is called "axial". Reflections involving four surfaces are called "tangential" and finally, reflections involving all six surfaces are called "oblique". The general expression for modal frequencies in a rectangular room is a well-known formula:

$$\omega_i = \pi c \sqrt{\left(\frac{n_x}{X}\right)^2 + \left(\frac{n_y}{Y}\right)^2 + \left(\frac{n_z}{Z}\right)^2}$$

Where: c is the speed of sound, c=344m/s, X,Y and Z are the principal room dimensions and n_x, n_y and n_z are positive integers (including zero).

The modes exhibit resonant character. They have natural resonant frequencies, bandwidths dependent on their damping factors and amplification Q-factors also dependent on their individual damping. Modal density varies in frequency. Modes tend to be widely separated at lowest audio frequencies and then come closer together as the frequency of interest is increased. In addition, the effects of low frequency modes on the source frequency response are heavily dependent on the position of the source and receiver.

For example: if a receiver is located in a pressure node, it will record no sound at all for a case of zero dumping factor for this particular node. In a typical listening room, exhibiting many modes, the frequency response of the source will be the vector summation of contributions of all modes, as well as the function of the source and receiver positions.

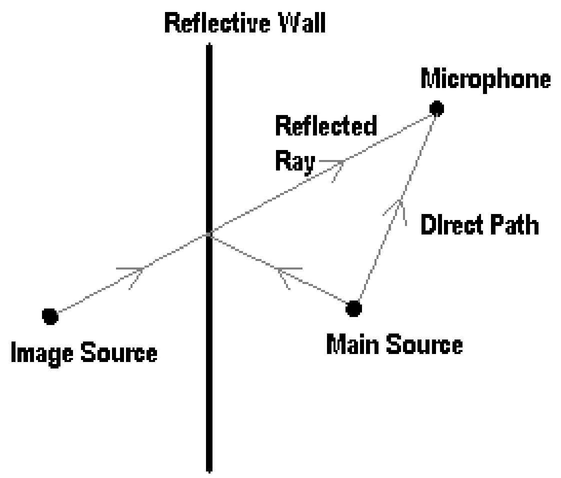

Fig 14.2 Image Model for single reflection

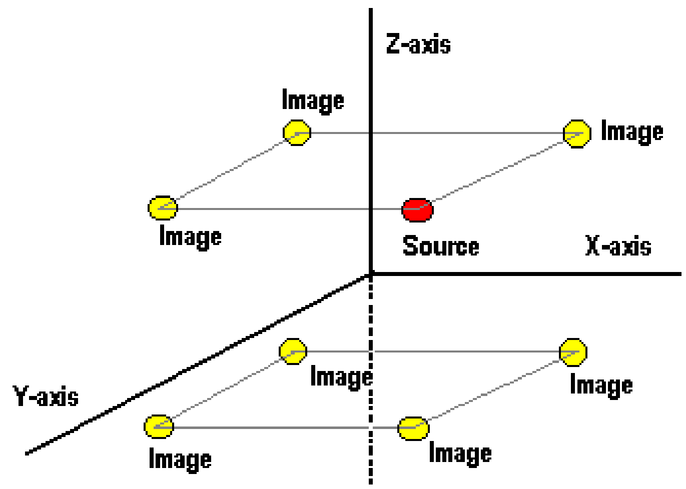

Fig 14.3 Image cluster (1 source, 7images) formed at a corner.

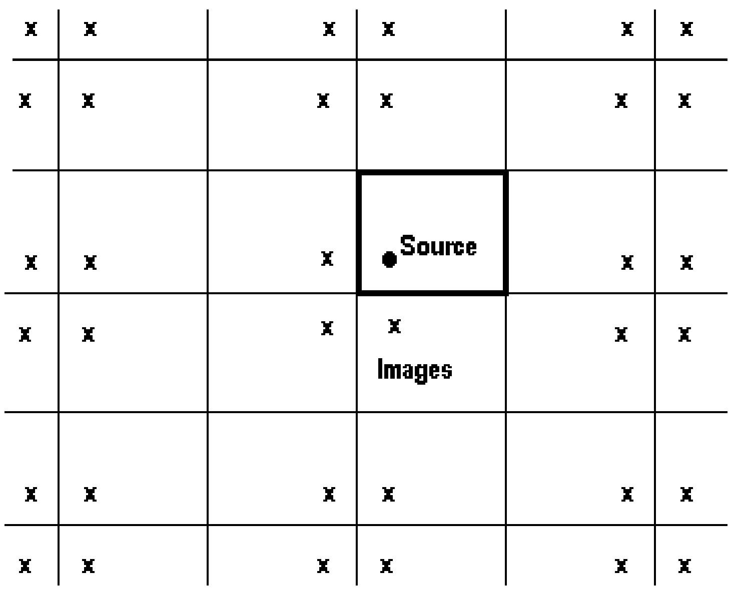

Fig 14.4 Positions of virtual sound sources on XY plane.

Using the Room Modes Screen

Please upload project file "test301.hif" into the program and select "**Room Image Method**" screen from the main menu - see Fig 14.1. The control dialogue box contains 3 scroll bars to position the speaker up to several meters from the corner and correspondingly, 3 scroll bars to move the "microphone" around.

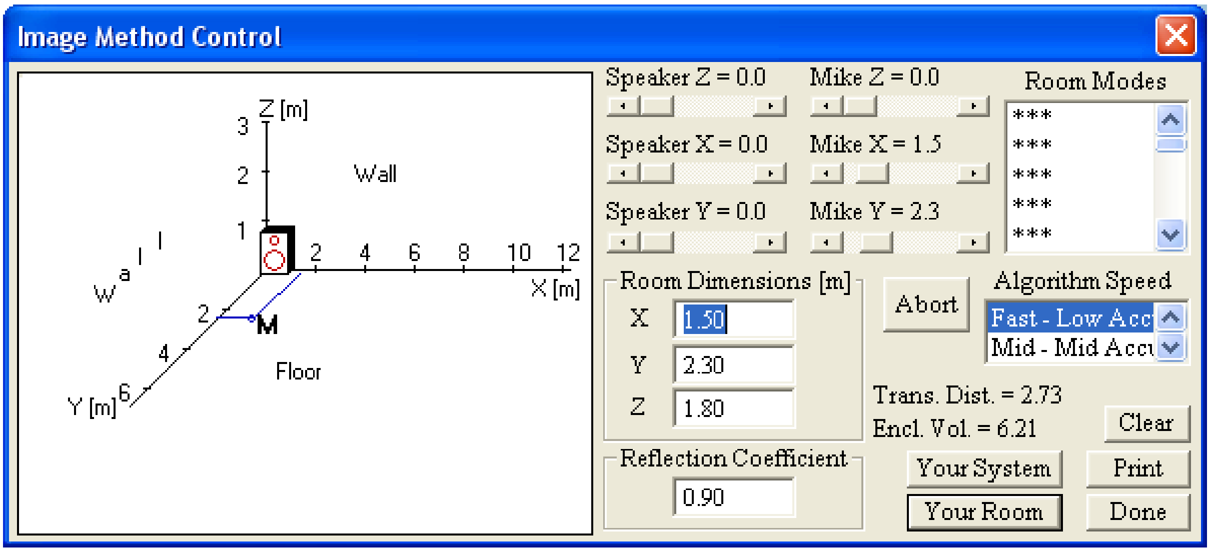

Figure 14.5. Image Method dialogue box.

Controls available in the Image Method Control box are quite simple:

1. **"Speaker Z"** – Controls location of the speaker along Z-axis (vertical).
2. **"Speaker X"** – Controls location of the speaker along X-axis (horizontal).
3. **"Speaker Y"** – Controls location of the speaker along Y-axis (horizontal).
4. **"Mike Z"** – Controls location of the microphone along Z-axis.
5. **"Mike X"** – Controls location of the microphone along X-axis.
6. **"Mike Y"** – Controls location of the microphone along Y-axis.
7. **"Room Dimensions X,Y,Z" –** Enter desired room dimensions in meters.
8. **"Reflection Coefficient" –** Accepts average reflection coefficient of the wall.
9. **"Algorithm Speed/Accuracy" –** Slow speed=high accuracy.

The dialogue box will display transducer distance and total enclosure volume. Pressing **"Your Room"** button will plot room contribution to the transfer function between transducers. Pressing **"Your System"** will add your system response onto the room transfer function. The **"Clear", Print" and "Done"** buttons do what the label indicates.

The loudspeaker can be position within X = 5m, Y=4m and Z = 3m distances from the room corner. The microphone can be located within X = 10m, Y = 12m and Z = 4m from the room corner. The horizontal frequency response is of 10-200Hz linear resolution. Room dimensions are editable from the three fields shown on the right-hand side. There is no built-in limit on the room size, but the screen is intended analyze small rooms. The room dimensions can be greater than graphical representation of the set-up shown on the bottom-left hand of the screen, however, the smallest room size is 1m x 1m x 1m. Pressing the "Plots" button invokes a floating menu, from which you can select various enclosures to be examined and also imported woofer and system frequency responses. The "Clear" button clears the plotting area. It is clearly observable, that the SPL frequency response of a loudspeaker placed in a confined space is far from being flat. The fluctuations in level are most pronounced at low end of the audio spectrum, where the modes are few and separate. The frequency response of loudspeaker will be further influenced by the actual location of the loudspeaker and the listener. The so called "corner location" allows the loudspeaker to excite all room modes - see Fig 14.1. On the other hand, locating the listener in the exact center of the room, further distorts the frequency response - see Fig 14.6.

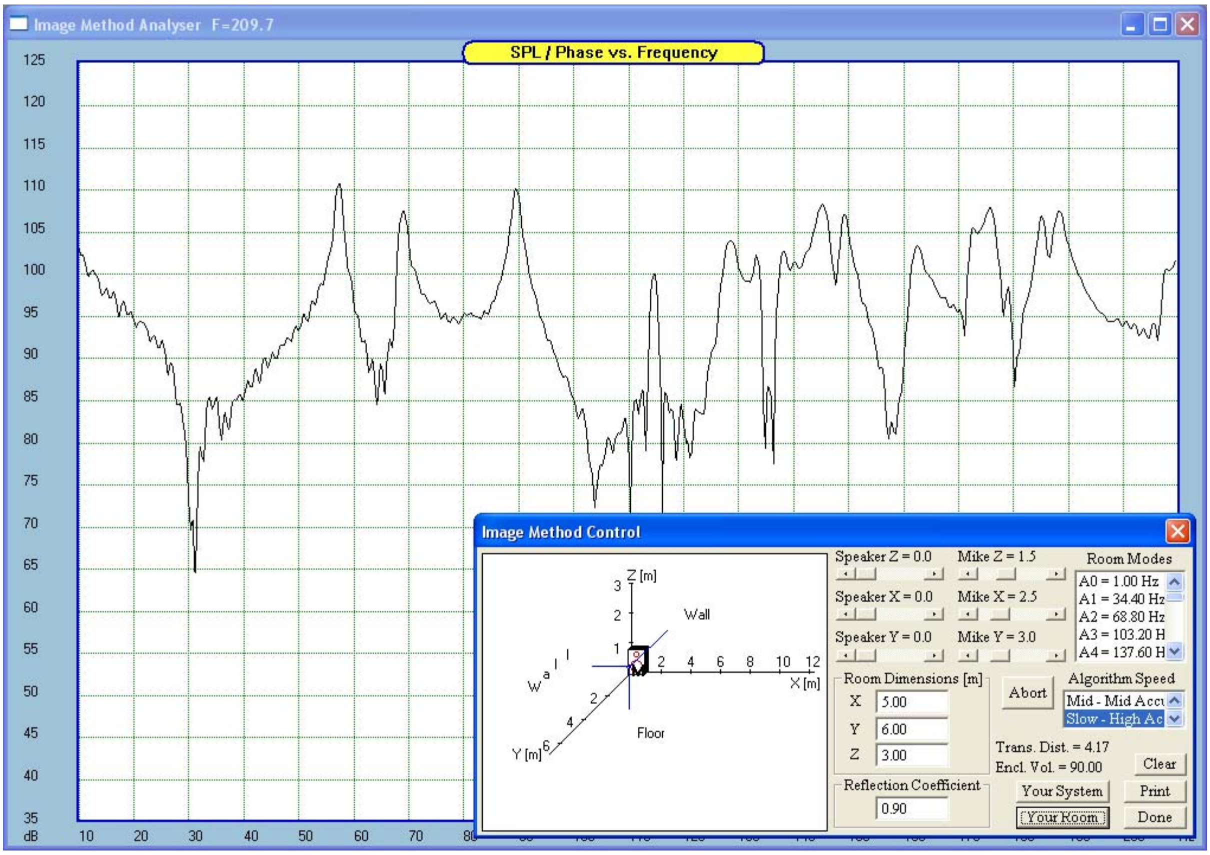

Fig 14.6 Central location of the microphone

Frequency independent, average room Reflection Coefficient (RC) can be entered in the corresponding data field on the bottom-right hand section of the screen. Please note, that entering the RC=0.01 (negligible reflection) corresponds to testing your design in the anechoic chamber. Indeed, when you use this low RC, the corresponding frequency response plot should look exactly as the one representing anechoic chamber. The RC=0.90 will produce highly distorted frequency response by contribution of the undamped room modes - see Fig 14.1.

The predicted "low end" frequency response is based on modal summation method. Calculations assume an empty rectangular room and the same absorption coefficient on all surfaces. The "real world" music room will rarely (if ever) comply with the above conditions. The room will contain furniture, the absorption coefficient of the floor and walls will be different and varied with frequency. The sum of all these factors will result is some differences between predicted and measured frequency response.

There is still quite a lot of sense in evaluating the predicted responses. The model is quite realistic in predicting frequency response deviations due to a particular location of the loudspeaker and listener. You will be able to select the best location for your speakers and plan the location of the listening area. For example: you will notice, that modal theory predicts large "dents" in the already highly irregular frequency response, when you place the listener or speaker in the center of the room.

Having worked with the screen for a while, you will find, that it is generally not desirable to place the loudspeaker or listener in the exact node of any given frequency, as this would create undesirable cancellations of the sound. The knowledge of modal frequencies and nodes' locations is essential in understanding your acoustic environment. This is why the FEM approach for predicting room acoustics in "non-rectangular spaces" is so successful. Even if the basic FEM implementation does not produce the frequency response, it will generate all modal frequencies and associated pressure patterns. All that is needed next is to review the pressure patterns and avoid placing speaker and listener in the nodes.

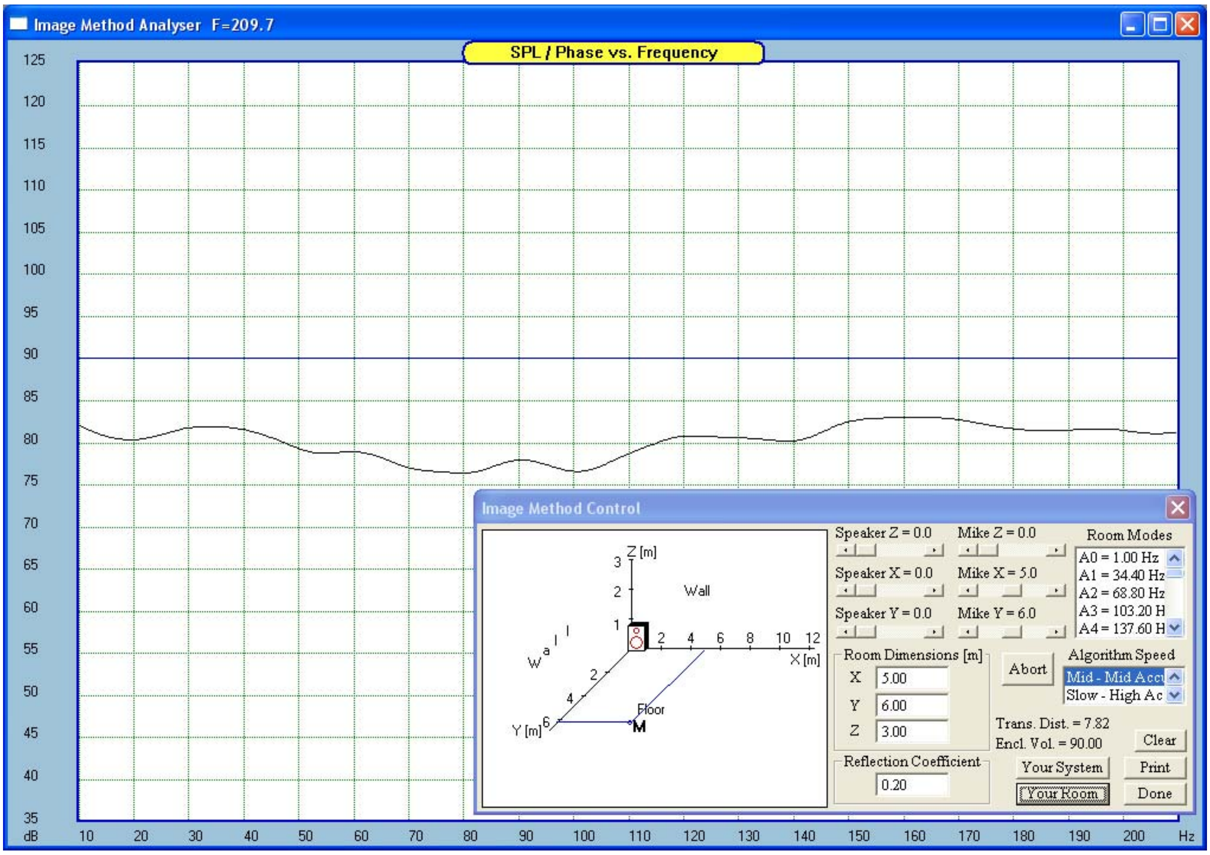

Figure 14.7. Low reflection coefficient.

Room Acoustics Background

At some stage you will arrive at the end of the loudspeaker system design process. It is probably fair to assume, that considerable effort went into designing and building the “perfect system”, possibly including anechoic flat frequency response from 20Hz to 20kHz and many other desired and unique features. There is one more challenge to consider on the way to actually enjoying the results: the listening room acoustics. Room acoustics issues are complex and for many of us the “learning factor” will be significant. Absorption, reflection and diffusion are the issues to be considered, together with ray-tracing and modal analysis. In the worst case, misunderstanding in this area may obliterate most of your design effort and a well designed system will loose much of its strengths. The room characteristics can never be removed from the listening experience and they must be considered if you are serious about what you hear. A crossover network, even if not perfectly optimized, will not introduce irregularities in the system frequency response of more than 3-4dB. Room acoustic may add as much as 15-18dB due to “room gain” factor and also cancel the acoustic output completely (well, almost) causing variations in the sound pressure up to 30-40dB. It is therefore highly desirable to understand tradeoffs and risk factors involved in evaluating the last component in the audio chain - the listening room. There are “rules of thumb” and common knowledge about placing the loudspeakers, but when it comes to showing the actual pressure distribution within the room and predicting room modes (resonances) for complex shape rooms, the situation is quite different. Among many “risk” factors, the following three seem to stand out: (1) wrong positioning of the loudspeakers, (2) wrong location of the listening position and (3) poor room acoustics. As a minimum solution to the above problems one would consider avoiding (1) and (2) and optimizing (3). The remainder of this chapter will proceed along this idea, keeping complexity to the introductory level.

Mid-frequencies Problem

From the theoretical point of view, below 300 Hz, the average listening room must be considered as a resonant cavity. In this frequency range, reflections result in standing waves and the room becomes a resonating chamber. Above 300Hz the Ray-Tracing Model becomes more useful. Here, we assume that the ray’s incident angle is the same as the reflection angle. In the home situation, the shape and size of the listening room are already fixed. For the purpose of reviewing the issues associated with room acoustics in mid-frequency range, we would like to start with a simple case depicted in Fig 14.8. Let’s assume that there are only two audio paths arriving at the test microphone.

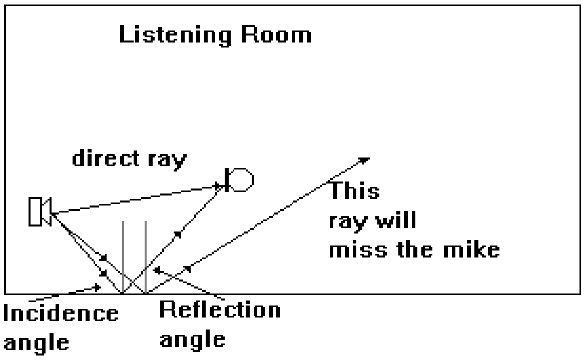

Figure 14.8 Incident and reflection angles.

If the reflecting surface is hard, there is no phase change between incident and reflected rays. Please note that the two angles are equal, so not every ray will reach the microphone. Figure 14.8 depicts the case, where two rays are combining at the microphone: (1) direct ray and (2) single ray reflected from the floor. The difference in length between these two paths is 10 cm. The 10 cm distance happens to be half of the wavelength (180 deg shift) of 1700Hz sinewave. If we add two sinewaves of the same frequency and shifted 180 deg, the result should be total cancellation. Indeed, this is clearly depicted in Fig 14.9. The first null occurs at 1700Hz followed by a +6dB peak at 3400Hz. At 3.4kHz the 10cm distance is equivalent to 360 deg phase shift, so both waves combine constructively. Next, another null occurs at 5100Hz, as the phase shift over 10cm distance equals 360+180 deg (phase reversal) and so on.

Adding More Rays

Adding more paths, each one delayed by a different amount, produces the effect depicted in Fig 14.10. In this example we have 8 rays added together - one direct and seven delayed. All delayed paths are 100 % reflected, there is simply no absorption in the reflecting surfaces. The resulting frequency response is heavily distorted. Similar analysis can be performed for any location inside the listening room. Are all the reflections necessarily bad?. Imagine that you have some perfectly absorbing material placed on all room surfaces and there are no reflections whatsoever. This is basically the situation in the anechoic chamber - the room is acoustically “dead”. Most people would discard such a room for listening pleasure. A solution seems to be somewhere in-between those two extremes.

Most distortions of the curve depicted in Fig 14.10 are attributed to early reflections, which would still have sufficient amplitude to compete with the direct ray. These are the first bounce reflections from the loudspeakers off the nearest room surfaces to the listening positions. Therefore these early reflections should be looked at first. You could determine the paths of the reflected rays using the graphical method discussed before. The reflection points are then covered with an absorbing rugs or acoustical tails - depending on the required absorption.

The next step is to look at the lateral reflections (reflections from the side walls). It is recognized that lateral reflections contribute to the image of the sound stage and spaciousness. It is therefore desirable to keep the lateral reflections only partially absorbed by the reflecting surfaces. This situation is depicted in Fig. 14.10, where three rays representing early reflections are 90% absorbed, two lateral reflections are 50% absorbed and the back wall reflections are 75% absorbed.

Professional literature lists absorption coefficients for a number of common building and decorating materials, so we advise you to consult the books on this matter. Absorption capability of the materials are heavily frequency dependent and are best in the mid-to-high frequency range. Porous materials, like foam and fiberglass, are very effective at mid and high frequencies, but loose efficiency at low frequencies. Therefore the above model and discussion is only valid for frequencies above 300Hz.

Please consult Chapter 15 for more information on Absorption Coefficients.

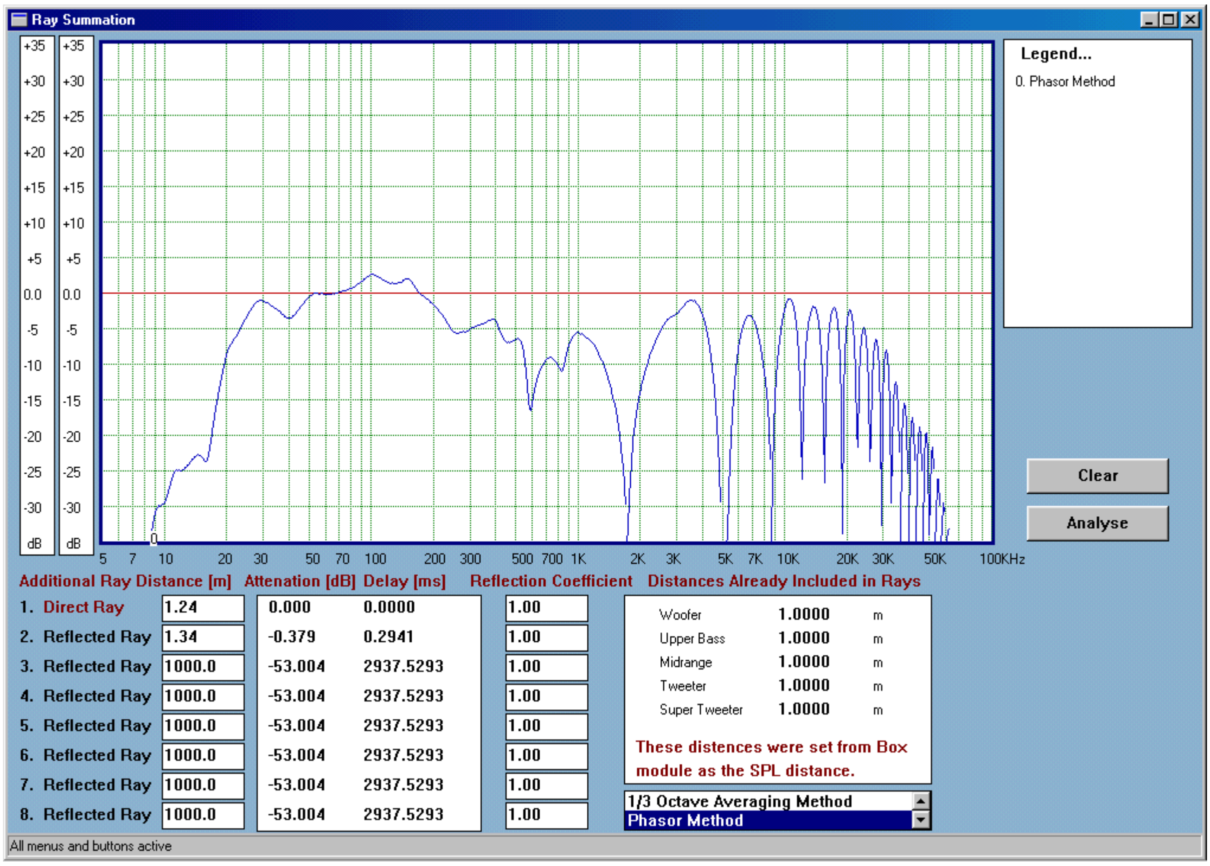

Fig 14.9 Two rays added – path difference 10cm

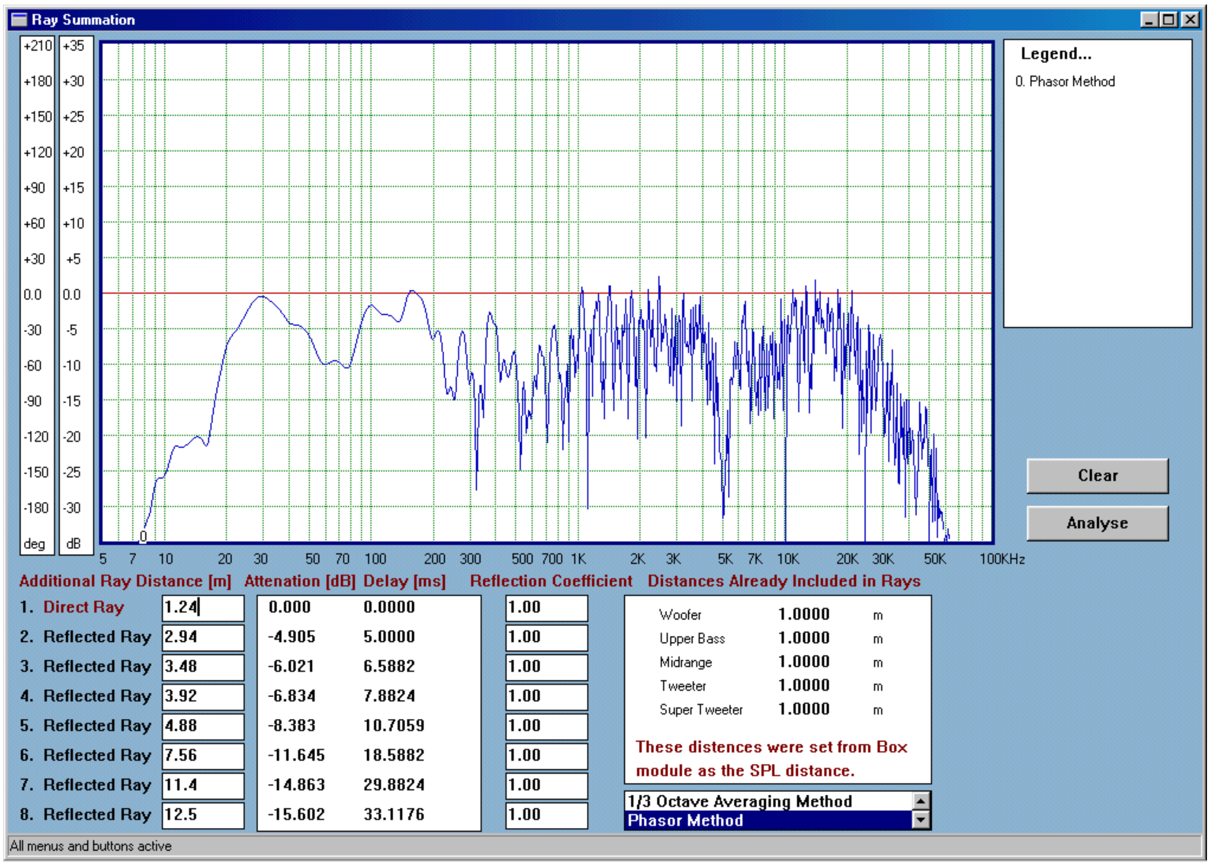

Fig 14.10 Eight rays added with various delays and 100% reflection

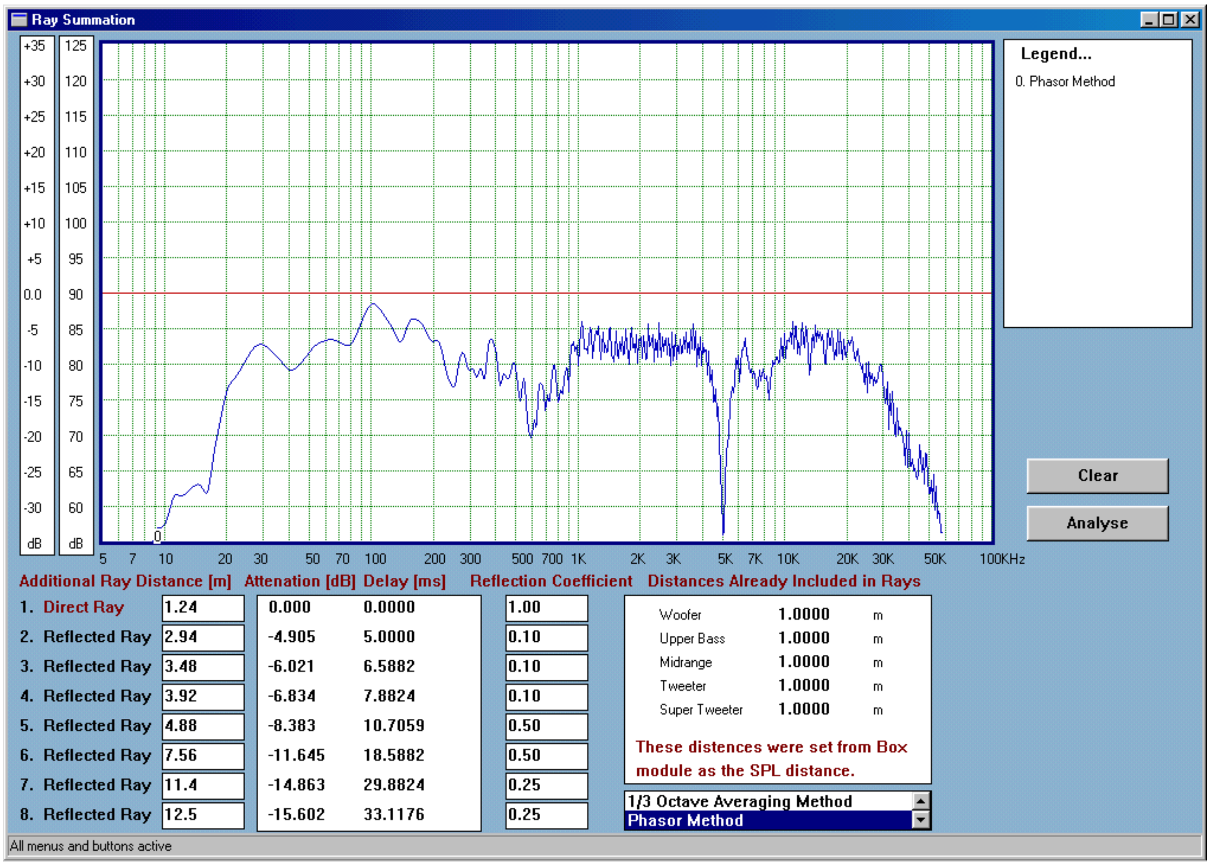

Fig 14.11 Eight rays added with reflection coefficients

The main purpose of the exercise above was to model summation of up to 8 audio rays on frequencies of 300Hz and above. Each ray may have different propagation path, with two parameters associated with it : (1) length, translated into attenuation expressed in dBs and phase change and (2) reflection coefficient, as the ray may be bounced off partially absorbing surface. The summed response of 8 rays is then plotted on the screen.

Room modes.

Listening room acoustics at the low end of the spectrum tends to be one big compromise, especially in small rooms. Standing waves are a fact of life in acoustically untreated rooms, and this will of course degrade the performance of a well designed loudspeaker system. Alternatively, installation of a loudspeaker system capable of energizing lowest room modes may well reveal that the room acoustics are less than perfect.

Room modes (natural resonant frequencies) occur across the entire frequency range. Professional treatment of the modes involves “bass traps” - typically a large absorbing element located in corners of the room. This idea may be readily applicable to a listening room in a recording studio, but for the rest of us it may be difficult to get all members of the family to agree to it. If this is your situation, there are a few things you may need to consider to avoid basic mistakes when analyzing low frequency issues of the home listening room.

Every room mode has associated acoustic pressure patterns with it. The pressure varies from 0 (null, no sound) to maximum (peak of the standing wave). When music is played in the room, the modes are excited accordingly and the pressure pattern changes and shifts dramatically within the boundaries of the room.

All modes, ie: axial, tangential and oblique, can be characterized by the following: (1) Bandwidth B - inversely proportional to the reverberation time: B = 2.2/RT60. Reverberation time depends on absorption, so the more absorption, the shorter the RT60 time and wider the mode bandwidth. (2) Decay - again, mode decay depends on the distribution of the absorbing material in the room and (3). Density - increases with frequency. Above 300 Hz room response smoothes markedly with frequency.

Let us now define the problem: you are the proud creator of a system capable of reproducing sound down to 20Hz, but when you sit down and listen to the sound, the low end around 20Hz is definitely missing. Assuming the problem is non-trivial, you are most likely sitting in the null of the pressure pattern for the 20Hz mode. The importance of knowing your room modes and associated pressure distribution is now becoming clear.

Mathematically, if the room has a simple rectangular shape an elegant formula can be used to determine room natural frequencies and some more work is required to determine pressure distribution. If the room has complex shape, the problem involves solving Helmholtz equation in 3-dimensions, looking for eigenvalues and eigenvectors. This process is handled very well by Finite Element Method (FEM) approach. If necessary, the FEM can be quite accurate. The accuracy depends on many factors and some of these are: the size of the element used, and order of approximating functions. The compromise on the other side involves computer processing speed and memory requirements. Therefore, in this example, we decided to accept errors of less than 5%, understanding that we deal here with an "approximate" method.