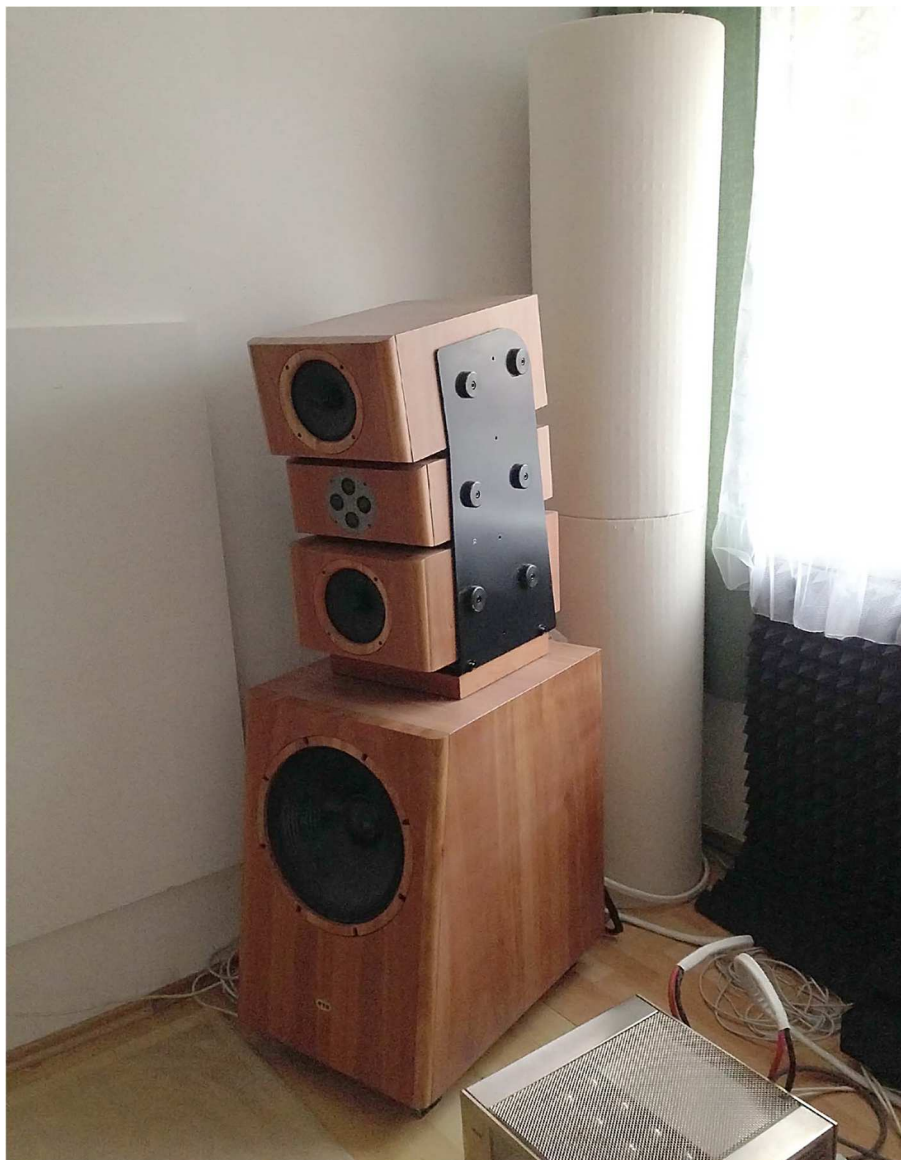




Fundamentals of simple acoustics Adjustments for high quality listening



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Infinite Search

I have met, during my life, many people who loved to listen to high quality music. Many of them have even invested a lot in their own equipment, and most of them are still looking for something better. Because they have experience from concerts and listening to live music, they know it's still not that. In the eighties, when I started with listening to audio, there were only gramophones and tape recorders. Many amplifiers were still with tubes and there were a lot of listeners of high quality audio. At that time, everything was analogous. Nobody was deprived of any data. Although we did not have as much high-quality material as we have today, the result was quite often relatively good. At that time, the capacitor dielectric was limiting (the quality of the insulation material inside a capacitor), so the problem was hum. There was no problem with noise at the time, because the tube amplifiers did not suffer from noise. But where the troubles were, was a quality of the magnetic materials for production of the output transformers. However, their use is necessary in tube amplifiers. Output transformers have greatly limited bandwidth of high power amplifiers. Tube amplifiers have had a very poor efficiency and, for reasons of high voltage demand, the tube anodes always required power supply.

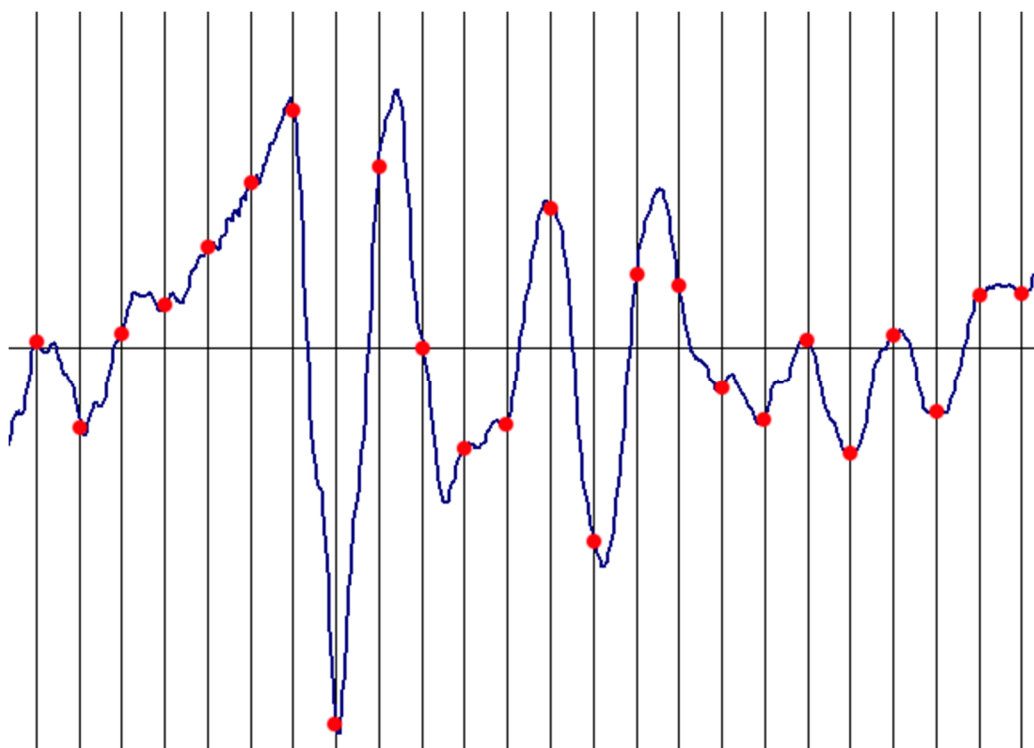
So there was no need to save on magnets needed for the speakers so much, so the only problem was the material of the diaphragm (mostly paper) and the material of the surround (the edge of the speaker membrane that holds the membrane on the speaker frame) and material of spider. Noise has begun to be a problem with expanding the use of commercial tape recorders.

The methods of acoustic measurements were very primitive and everything was measured point by point manually, and some things could not be measured at all. Everything evolved in a direction that did not bear audiophile malice. People practically stopped production of tubes, everything was replaced by transistors. Manufacturers were focused on product minimization, battery power, portability and price. With the tape recorders, tape motion slowed, and its width later diminished. This increased the level of basic noise and increased crosstalk from the right to the left channel, because the tapes were already very narrow. Only production of cassette tapes began, classic tapes ceased to be used and audiophiles suffered. Fortunately, the turntables survived. Development of cassette tape players hit the physical limits of magnetic strips and the world later left them. With the arrival of CD players, digital time has come. Whoever remembers it knows that the beginnings were difficult. Today, we know that the originally agreed 44.1kHz and 16-bit sampling was not bad, but the technique at that time had no experience with Digital Signal Processing for Digital Analogue Transmission. The biggest weakness of the agreed sampling method was the sampling frequency of 44.1 kHz because the audible bandwidth of the human ear is wide enough, and if the physical sampling limit is two samples per period, the limit for the transmitted frequency is 22.05 kHz!! The tapes ceased to be used and audiophiles suffered. And because everything that is produced above 22.05 kHz does not belong to the audio signal, it is necessary to solve a filter that would release 20 kHz and suppress all over 22 kHz. This is no longer possible to implement the analog filter, so digital filters have started to be used in all CD players. Who knows the filter theory knows that if we want to limit the amplitude (level) to 20 kHz, our analog filter will cause that the phase will bend already octave before the cutoff frequency, that is 10 kHz!!

(Changing the phase means moving the signal over time, so to send a signal to such filter cause, that the signal going through LP filter will come out at a different time than the signal going through HP filter!! The final acoustic signal is a combination these two, time shifted signals and it will fundamentally change the color of the sound. Audiophile was hurt because he could not hear it, but the commercial industry was making money.



Because manufacturers assumed, that the main problem was in sampling speed, it did not take long time and new formats such as DVD audio, SACD and so on came to the world. Sampling rate increased to 96 kHz, so the physical sampling limit allowed to transfer up to 48 kHz. Even own filter had a simpler task because it had to release 20kHz and eliminate everything above 48kHz. But you are wondering, it still did not play as it should! Fortunately, the world of technology has evolved very rapidly, and computers have come up with incredible possibilities to use them. The arrival of high-quality sound cards to PC also caused arrival of computer programs for acoustic and electronic measurements and it all began to be revealed gradually. It has been discovered that Jitter is the most basic problem in digital audio processing. Jitter is a variable that defines an error at the time the digital signal is to appear (i.e., the unstable clock frequency of the oscillator). What does it mean for a DA converter? Anyone familiar with the analog-to-digital converter knows that the signal is so-called sampled, it means that the level is measured at that moment, converted to a number (At a given number of bits - zero and one) and the time and the level are recorded.



You can see red dots in the picture, the sampling points. The vertical lines show the sampling time - the so-called clock frequency. The digital analog converter has exactly the opposite role, and its output is the process of linking the red points. Jitter is the shift of the sample position (red dots) in the horizontal direction - that is time. The level (vertical direction) is not a problem, because even 16 bits are very precisely defined. 16 bits means 65,536 levels. The output of LF signal from the CD player is usually 2V, from which 65 536 we get steps per 30 μ V. Such a value cannot be measured. Because time is only divided into 44100 steps and we can use from them only one half, because we need at least two samples per period, we're doing much worse.

$$1 \div 44.1\text{kHz} \div 2^{16}; \div 2 = 173 \text{ (ps)}$$



Who knows about the theory of digital filters used in the CD player then knows that the DA converter may have multiple oversampling, which multiplies the requirements for jitter even by the number of oversampling.

Modern devices can measure Jitter very well and very quickly it turned out that Jitter in standard commercial CD players moved more than the order above!! Who wants to read more, I suggest finding a "non-oversampling digital filter-less DAC Concept" article from Mr. Ryohei Kusunoki on the web.

They started looking for solutions and the quality went up sharply.

Another amazing thing is the internet. Today, everyone has the choice of what quality he wants to listen to, and can buy virtually anything from 16 bits / 44.1 kHz to 192kHz / 24 bits and even higher today. Nowadays, audiophile has a decent notebook with internet, a high-quality DA converter, amplifier and loudspeakers.





My story

It is true that I started a long time ago and I tested for a long time and was looking for a solution, why it did not play as it should. My first experiences started in high school where we tried to build my first amplifiers with my classmates. At that time, our educator, who supported us in these activities, helped us because it was not easy to steal such a huge number of teenagers who could not go almost a week out of the school building and boarding school. Our biggest problem, however, was always the lack of know-how. At that time, we did not have internet and the knowledge of my classmates and our educators were always endless in what they taught us at school. Although I have studied communication technology on both, my high school and my University, I learned up to several years after school that the most important thing that is inevitable for true listening has never been taught us. So, we are trying to build the amplifiers with the highest possible power, with the lowest distortion and the highest signal-to-noise ratio. Today, I know that this is just one of many other conditions necessary for faithful listening in small (home) listening spaces.

After velvet revolution, a gate of unrealistic possibilities opened and in the nineties, began a huge boom with the import of all possible audio components. There were also many Hi-Fi studios, Hi-Fi magazines, lot of traveling, visits, listening, shopping and trying everything, but according to what I myself experienced and saw with my friends and other friends of faithful reproduction and other candidates for faithful listening, everything have been turned down and the whole market has almost disappeared. I have limited my activities also and devoted myself to full employment. And since I worked for various managerial positions for more than 20 years, I had less time for my hobby. It took me a lot of time, because I could only listen to the audio during some weekends.

Gradually, I built and tested my various designs, and around the year 2000 I began to try using the available acoustic measurement programs. Finally, I understood why we were still counting different transformations at school (mainly Fourier), and I myself tried how we can make very accurate very fast measurements with the growing possibilities of standard personal computers. Because I thought the weakest part of my acoustic chain were laoudspeakers, I devoted myself fully to their design, production, testing and measurement. This attracted my neighbor and he already knew Mr. Novák, he introduced us and began a long-term cooperation on the development of the whole acoustic chain. Although I was equipped with a very good device for various electronic and acoustic measurements, I was very surprised how important part of our body are our ears!

Mr. Novák had many years of experience as a sound engineer and what I heard at his place have taken my breath. I quickly realized that high quality listening is not just about high power, low distortion and high signal / noise ratios. My technocratic (school-influenced) approach has changed very quickly. Our multi-year collaboration has resulted in a complete solution of the whole acoustic chain from the source of the acoustic signal to the listening room. We have tested many different products, components, cables, theories, and everything else to confirm that physics is still valid and that our great-grandfathers already knew how should high quality sound look like. Unfortunately, the commercial market is constantly escaping compliance with physical laws, only to save and make the most of it.

Because feelings of faithful reproduction are amazing, and because much of my life has already fled between my fingers, I have decided to make use of my life experiences and to help listeners to understand what is needed and to allow them, to listen at my place, how it can play, or compare their audio components to mine. I will teach them what listening room to choose, how to edit it and what to do it. Today's possibilities are incredible, but it's always necessary to hear it at first! I burned myself a few times when I believed in a beautiful conversation. Let us know, and we can make a deadline and you can listen to what you want.



Basics for high quality listening

The basic condition for faithful listening is a suitable acoustically treated room, a good source of acoustic signal (uncompressed music file, DVD audio, SACD, even CD and of course LP), good quality whole acoustic chain (since this is a serial connection, the resulting quality is proportional to the weakest part of the acoustic chain, so that a poor signal cable is enough and it will not play well) and the right listening place. Because we are talking about stereo listening, there is only one place in the listening room for faithful listening!

It is necessary to realize that the acoustic signal comes from the source directly - direct acoustic signal and indirect reflections from all surrounding walls and obstacles. An indirect signal is always slightly delayed compared to a direct signal because the sound moves at about 300m / s. So, if the reflected signal must travel 3m longer to your ear, it will be 1ms delayed compared to the direct signal and will have a lower amplitude as the acoustic signal level decreases with the second magnitude of the distance. But our brain is very smart and knows how to compare these signals coming into his ears and can create a picture of the listening room accordingly. If the reproduction of this record is to be faithful, it is necessary to ensure that all the direct and indirect reflections for both ears (in case of stereo) are recorded correctly in the recording room. When reproducing such a high-quality acoustic signal, it is necessary to ensure that listening room itself and the entire reproduction acoustic chain do not affect this signal at all. For faithful listening, both direct and indirect reflections in the recording room must go through the two microphones into the recording, and then reproduce from two speakers and arrive in your ears as if you were sitting in the concert room while listening to live. Who has only a little awareness of how the whole acoustic chain works, so he knows it is not easy at all! The human ears, in junction with a brain, have incredible possibilities mainly in the dynamic range. Although the intrinsic frequency range is not very large, let's say 20Hz to 20kHz, the dynamic range is over 100dB. To be able to imagine it, 20 dB is a difference of level one to ten, 60dB is one in a thousand and 100 dB is one in a hundred thousand!! Since sound is a very complex matter and is always composed of many frequencies of different amplitudes and phases (it means size and shift over time), it must always be transmitted undistorted not only in amplitude but also in phase!!! That's the biggest problem! If you change any part of the acoustic signal, even if it is a harmonic frequency (multiples of the basic acoustic signal frequency) at levels of 40 dB below the basic harmonic (that is 100 times less than the basic harmonic), you will change the resulting sound color. Every lover of live music knows this. But this is the basic principle of compressing the acoustic signal (e.g. mp3). The goal is to save on the amount of data, i.e. to have the smallest size of data files, to be able to be transmit it with as little as possible requirements for transmission paths. For faithful listening, use uncompressed music files or acoustic recordings, i.e. LPs or tapes. Today there are already commercially available uncompressed music files with sampling rates of 192 kHz and 24 bits, and sometimes there is even a recording at a resolution of 385 kHz / 32 bits.

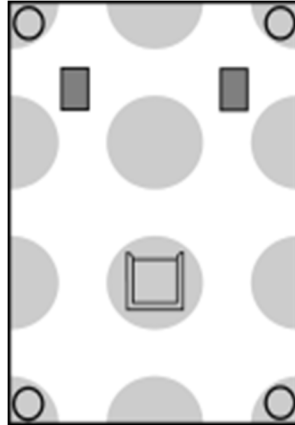
Listening room

Many listening tests and, in turn, the experience gained show that the entire system is actual series of individual elements of the entire audio chain. By skipping the entire recording phase, which is mostly done on a professional level, we will go to the audio source that we have available and would like to listen to it in the highest possible quality. As it has already been said, this is a serial connection of individual audio



components and from that follows that the resulting quality depends most on the weakest element of your audio chain. Every part of this chain has some influence and, according to many listeners and even your own experience, you will soon realize that the more you approach the end of the audio chain, the more the audio elements have more influence! Nowadays, if we forget compressed formats (like mp3s, etc.), the sources of audio recording are very high, and the parameters of distortion and signal-to-noise parameters are confirmed. If I choose a CD player, its low-cost version will have better standard parameters than a much more expensive power amplifier. But even such power amplifier is usually cheaper than loudspeaker system and, ultimately, its distortion is a few orders lower than the distortion of actual electroacoustic transducer. But it's still not the element in the acoustic chain with the greatest effect on the resulting quality. The listening space has the greatest influence on faithful listening. Since the dimensions of home listening rooms are small in terms of sound wavelengths, for their treatment are valid different conditions than for large listening rooms, such as concert halls or even outdoor spaces.

In the small listening room, there are many reflections and many standing waves. Direct reflections from the nearest walls cause mostly distortion at mid and high frequencies, due to the fact, that a direct acoustic signal from the loudspeaker system comes to your ear, but also a direct reflection signal from the walls, floor, or ceiling that is only several decimeter or meters longer. Even though the acoustic signal level drops with a square of distance from the source, this signal is only slightly lower in acoustic levels and only a few milliseconds delayed than the direct signal because its path to the ear is only a little longer than the direct signal path. These two acoustic signals are added together and thus there is a large distortion in relation to the original acoustic information from the electroacoustic transducer (loudspeaker). (To be able to imagine 1 ms of delay, imagine that 1 kHz (the middle frequency, which you can hear when the TV is out of the broadcast and you only see a stationary rounded picture and hear an unpleasant whistling - it has a frequency of 1 kHz) has a period of 1 ms, and the sound speed is 330 m / s. It means that, if you have a half ms delayed signal, i.e. the indirect acoustic wave has a $33/2 = 16.5$ cm longer way from the surrounding speaker surface and the final amplitude is then the difference between the amplitude of the direct and the reflected signal so they are completely subtracted !!). Since there are much more reflecting surfaces in the small listening room, the resulting acoustic signal received by the human ear is even more distorted. Between the walls and between the floor and the ceiling, during reflections occur so-called standing waves that cause the given frequency corresponding to the distance between reflecting surfaces have a different level at each room location, meaning that at the point where the vertical amplitude has the biggest value you can hear only this frequency, and at the point where the vertical wave has the smallest amplitude, there is practically no sound at all. The dimensions of the listening space must therefore be suitably selected and suitably adjusted acoustically, to avoid undesirable acoustic reflections from the solid surfaces, i.e. the walls, the ceiling and the floor. The layout of such standing waves is shown in the picture below. High level basses are marked with gray areas.

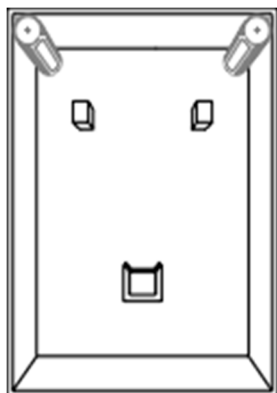


Because standing waves occur in length, in width, and also in height, it is necessary to choose the room dimensions (if possible) so that the unwanted effects of standing waves are not supported! For example, if the width of the room is exactly two times the height, the effects of standing waves in width and height will be fully supported and the result will be even worse. If you have a possibility, choose your listening room to have approximately the dimensions according to the following room dimensions:

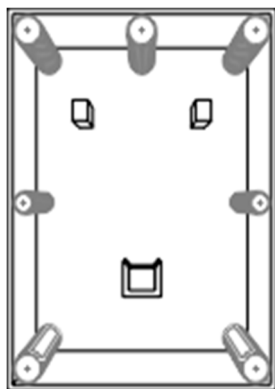
1 : 1,14 : 1,39 (height : width : length)
1 : 1,28 : 1,54
1 : 1,60 : 2,33

This means that, if we consider that the room height is 2.7 m, then the width would be $2.7 \times 1.14 = 3.1$ m and length will be $2.7 \times 1.34 = 3.7$ m. By that you will save yourself many unpleasant moments and especially the cost of acoustic adjustments to the listening room.

The front side of the room is where the speakers are located. The location of the speakers should be symmetrical with respect to the front walls of the listening space. The surface of the walls should be relatively hard without open doors or corridors, to avoid another release of the sound pressure from one speaker as from the other. The speaker systems are usually spaced apart, usually 2 to 4 m. The room width must be reasonably larger, typically at least 4 m, otherwise the speakers record a certain degree of acoustic load, so-called megaphone effect, which changes the efficiency of the speakers in the lower frequency band and low bass are then too strong.



In the figure, you can also see the basic layout of the tube-traps in the corners of the room behind the speaker systems (high acoustic dampers to the lowest frequencies). Since at the corners there is always the highest level of low frequencies and the corners in the corner are still aesthetically manageable, it is the most important and most effective step for limiting standing waves at low frequencies. You get more efficiency, when you put the tube-traps in all four corners of the room.



The next step may be to add five more tube-traps as shown in this figure.

Tube-traps are manufactured with different diameters. The higher the diameter, the lower frequencies it can damp.

Approximate ranges of working frequencies are in the following table:

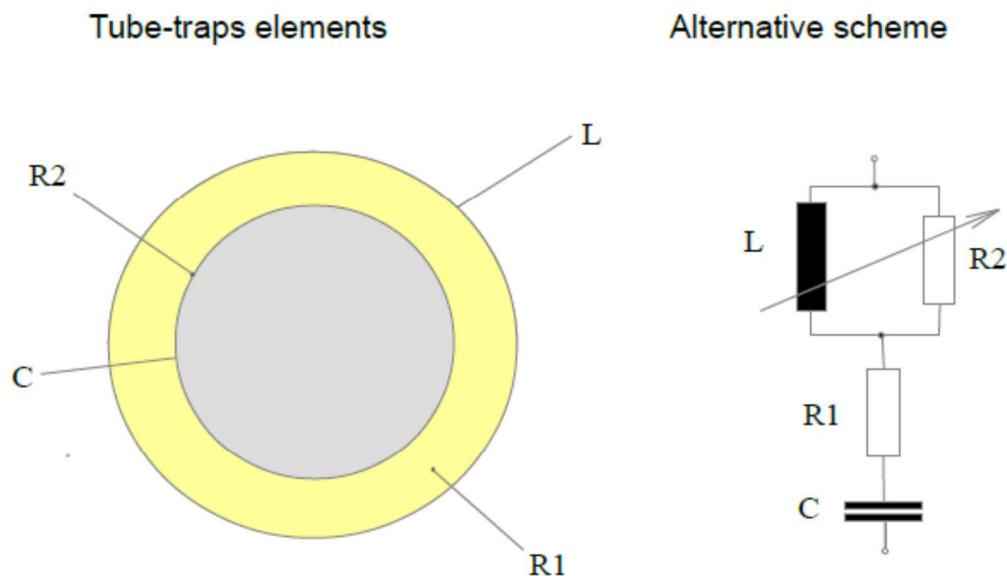
Tube diameter	Rolloff frequency	Crossover Frequency
230 mm	110Hz	400Hz
280 mm	90Hz	400Hz
330 mm	70Hz	400Hz
400 mm	55Hz	400Hz
508 mm	40Hz	400Hz



How Tube-traps Work (A Little Theory to Explain):

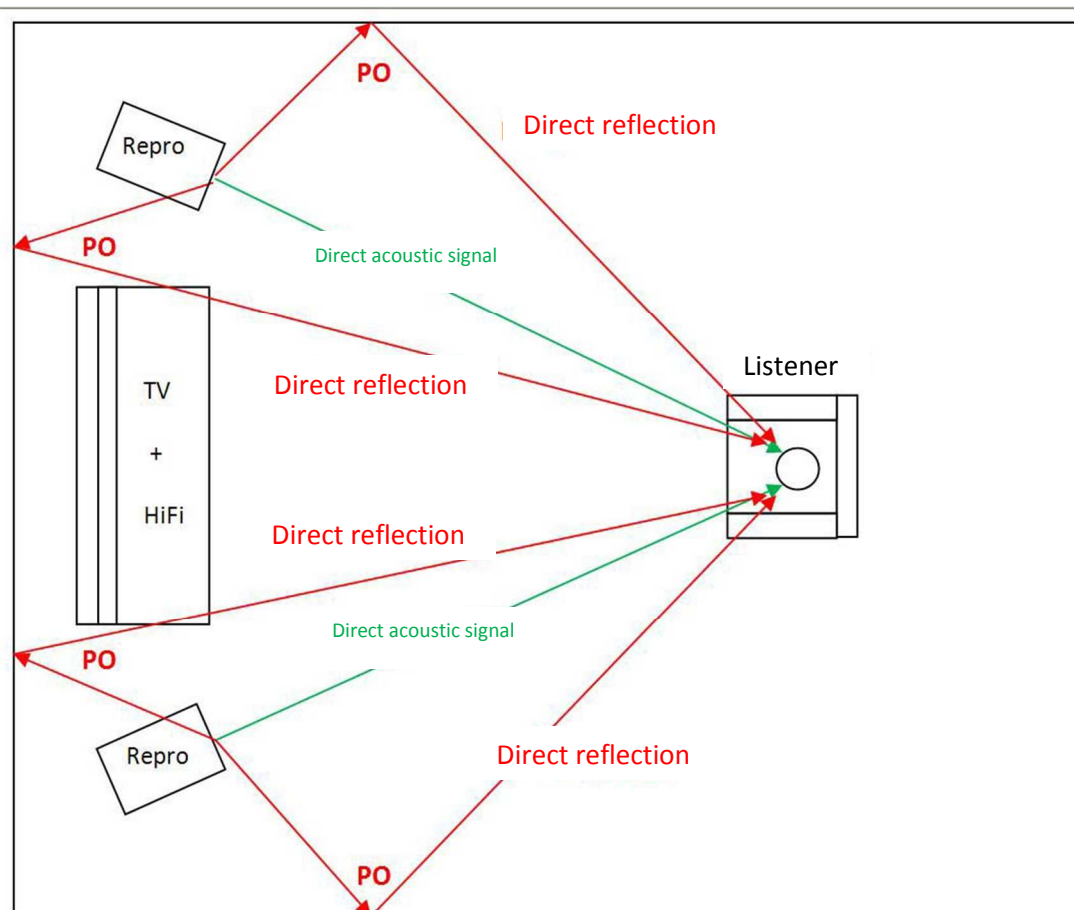
The Tube-trap is the first and the best bass trap in the audio industry. Unlike other bass traps, the damping physics is based on the RC circuit. The acoustic capacity (C) is given by the air chamber inside. The bigger it is, the more efficient the bass trap is at low frequencies. The acoustic resistance (R) is a DC impedance coupled to the radiated (LC) impedance of the freely passing sound wave.

These bass traps are therefore very effective. Another unique feature is the adjustable diffuser part of the Tube-trap. The half of the Tube-trap circumference is provided with a reflective foil that produces acoustic inductance (L) and reflects the frequency of the heights, depending on where it is rotated.





However, middle and high frequencies must be addressed differently. In the figure below, you can see which basic acoustic signals of this frequency range will reach the ear of the listener. However, there is also another and very large number of reflected acoustic signals from all other surfaces. But these are much lower levels, because they must fly much longer way. Because their level is limited by the square of the distance, and the middle and high frequencies are much more attenuated by reflections from different materials than the low frequencies, we will not deal with them and focus on direct reflections (POs) that have a major influence on reproduction quality.

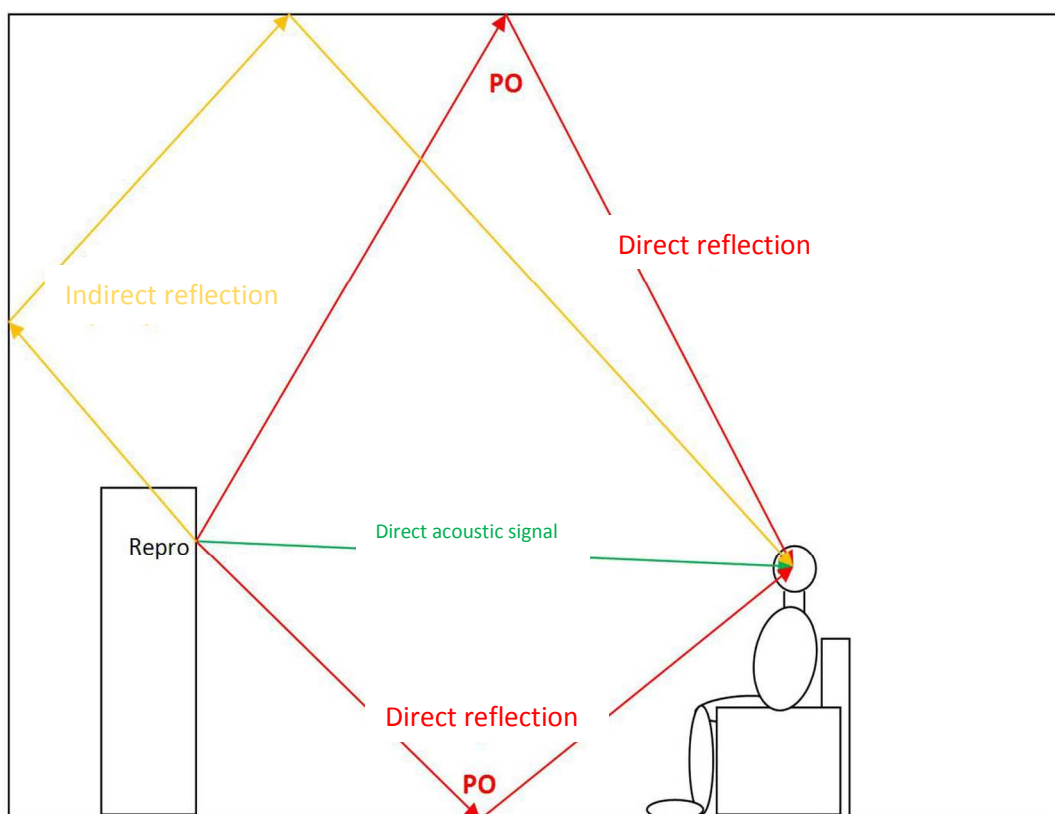


In the above figure, you can see the optimal layout of the elements in the listening room from the floor plan and the side view below. Very important is the damping of direct reflections at PO points!! Looking from the side, it is necessary to realize that the loudspeakers are two, so the damping of the points of direct reflection for each box must be separately solved.

Every listening space is different and its equipment can also change. Still, basic physical principles are valid for you at home as well as for all audiophile listening compartments.



Placement of direct reflections when viewed from the side:



Acoustic damping causes everything that is in the way of an acoustic signal, i.e. curtains, carpet at the reflection point, upholstered furniture, etc. In PO points, however, it is necessary to use material that has very good damping for the acoustic signal and its appearance is not repulsed, because listening space is usually placed in the living rooms. We are using acoustic plates, which are mostly made of mineral wool by professional manufacturers and are smoothly treated and white so they do not disfigure the listening room.

A very good solution for the acoustic treatment of the listening space is the use of a suspended ceiling with acoustic panels, filling the air gap over the ceiling with a mineral wool that has good acoustic damping. It is usually enough to hang the ceiling 30 cm below the ceiling and install 20 cm of mineral wool insulation above the ceiling. An example of such a solution to the ceiling damping is shown in the figure on the next page.

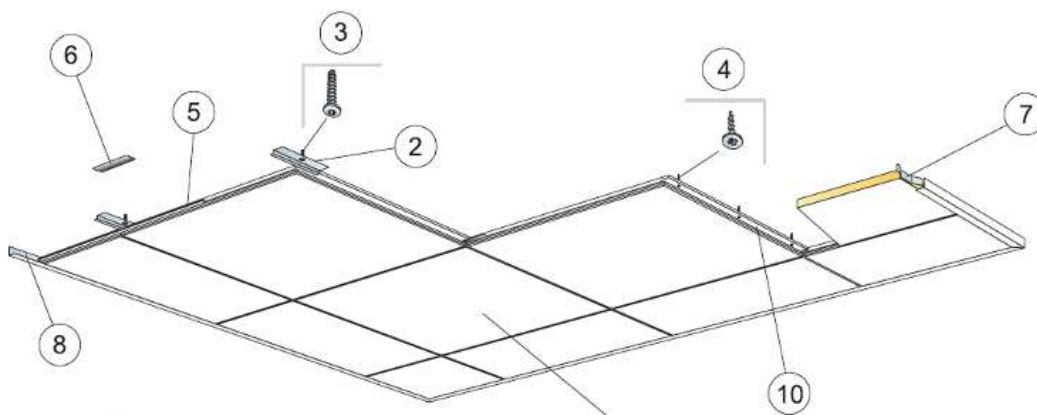
Be sure to use at least the carpet at the point of the direct reflections of the two speakers from the floor! This will greatly clear the recording information. This of course applies to all places of direct reflection!!

To be healthy, be aware that acoustic comfort is not only important to audio

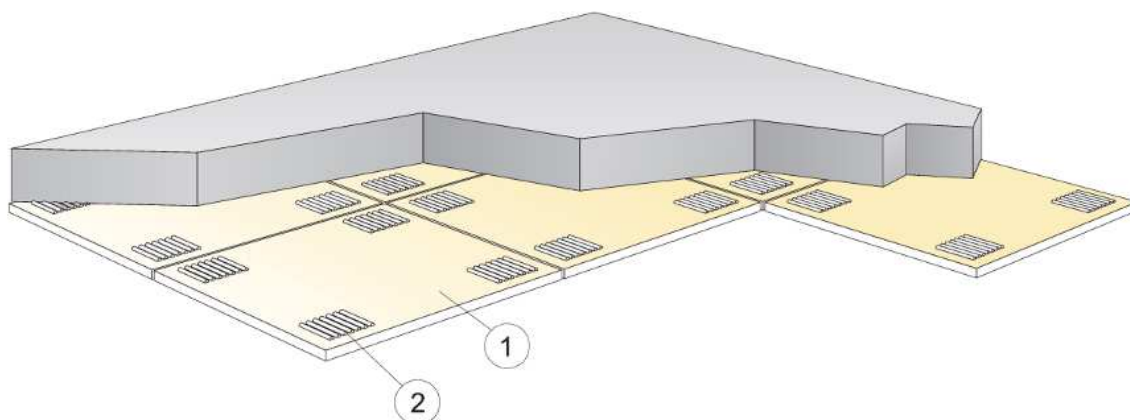


listeners, but is important to all people staying longer in a given space, because bad acoustics have a very negative effect on the psyche and, of course, human health.

Design of hanging



Acoustic insulation above the ceiling to increase efficiency at the lowest frequencies:

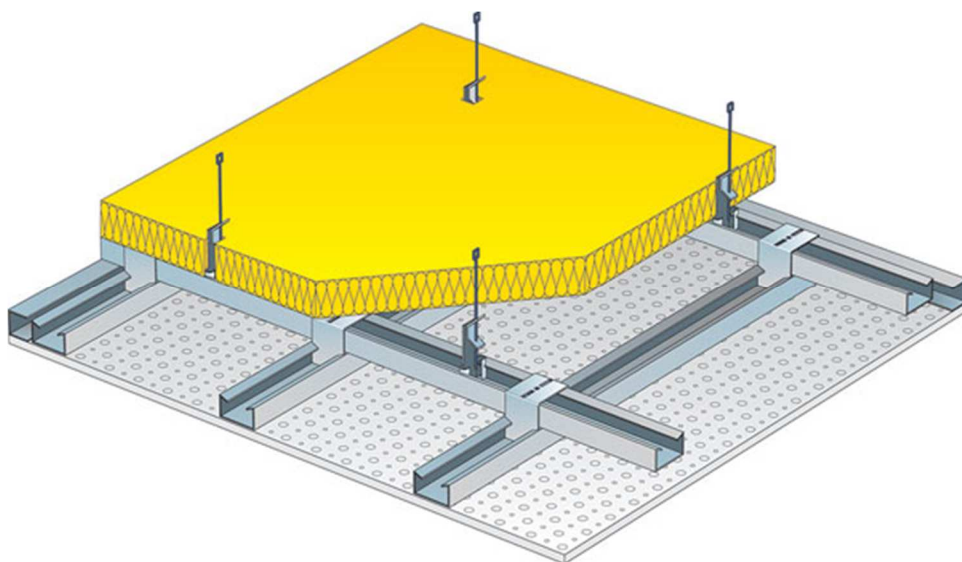


Practical appearance of the ceiling after finishing:





The manufacturers of these panels are more and it depends on who you prefer.



In the figure below you can see the insertion of one layer of 10cm mineral wool. I recommend, to put there two layers. The bottom of the insulation must always be covered with a vapor barrier, which prevents the insulation from dusting down. Then acoustic panels are hanged up.





The spots of direct reflections on the side walls can be solved by gluing a 40 to 50mm thick acoustic plate as I have in my listening room or the whole wall can be damped. It is aesthetically perfect, but it is considerably more expensive in terms of cost.

You have seen my minimal damping on the front page of this E-book.

Here you can see how you can adjust the listening room very well, which can still be used as a living room.



Cooperation

Since there are virtually no limited possibilities to buy anything anywhere in the world today, we offer cooperation in selecting and purchasing your components at a reasonable price / performance ratio and possible further adjustments for gradual upgrading of quality in incremental investments. Not everyone has possibility to get what he wants right now.

Nowadays, both DAC converters and tube amplifiers can be purchased at a very reasonable price and can be gradually adjusted. We offer many solutions for such successive steps. Visit our website and read it carefully and see that you can find a reasonable solution.

Alternatively, arrange a visit to our listening room and come and listen to how it plays and how you can improve your own audio system yourself.