HiFi PiG MAGAZiNE

An Introduction to Small Room Acoustics & Room Treatment Part 1

by GIK ACOUSTICS

A HiFi Pig partnership-promotion with GIK ACOUSTICS

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Front cover image supplied by GIK Acoustics courtesy of Richard Fox

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ABOUT GIK ACOUSTICS

GIK Acoustics was founded in 2004 in Atlanta, GA, by Glenn Kuras, President, who recognized the need for an acoustics company that could fit the demands of a variety of rooms from home listening & recording rooms to world-renowned facilities like Abbey Road Studios. Glenn is an industry leader who holds a patent on his acoustic design and as a pioneer, Glenn continues to introduce new and innovative products. Glenn has been extremely proactive in disseminating educational articles on acoustics and helping to make the art of small room treatment accessible to all via videos articles and forum presence. David Shevyn, Managing Director GIK Acoustic Europe opened the European branch in partnership with Glenn in 2009. David is heavily involved in the Research and Development of GIK Acoustics products, lectures on acoustics throughout Europe; from Munich to Milan, Dublin and London and contributes articles on Small Room Acoustics in several Pro Audio and Hi Fi Magazine and from 2019 is a regular contributor and columnist on the topic for HiFi Pig magazine in the UK.

OUR PHILOSOPHY

Since the inception of GIK Acoustics in 2004 (and later GIK Acoustics Europe in 2009) we have guided the company with the following principles:

QUALITY – We use only the highest-quality materials that will not sag or leak over time, then we build every panel by hand employing a unique two-frame system.

VALUE - Dollar for dollar, GIK Acoustics' products absorb more sabins (sound) than any other product on the market. We provide our customers the most cost effective solution to make every space sound its best.

INNOVATION - GIK Acoustics is also proud to have launched creative unique products such as: the multifunctional PIB (Portable Isolation Booth), the decorative and effective Impression Series, and the striking, precise Gotham N23 5 Quadratic Diffusor.

SUPPORT - We start by helping customers determine the best products to treat their space with our free, expert Acoustic Advice. Our well-respected design team has years of experience and are available to provide assistance not only with product selection but product placement within your room.

INFORMATIVE - When it comes to improving the sound quality in a room, many find the science to be overwhelming and the task of DIY'ing acoustic treatments to be daunting. Our solutions are based on the science of acoustics and one of the key dynamics to our success lies in demystifying sound dynamics and room acoustics. We are firm believers in education and our website is full of Educational Articles and Videos.

ENVIRONMENTAL - It's our commitment to use environmentally safe materials for all our products. We use ECOSE absorption material, eco-friendly wood and we offer customers recycled fabric options as well. Greensafe means GIK Acoustics products are friendlier on air quality, the environment, and your budget.

It is our belief that given the right education, products and guidance that everyone is able to transform their sound of their room in a positive way. We take a holistic approach to acoustics and our in depth understanding of the topic and the practicably of transforming small room acoustics has allowed us to introduce an award winning array of products to meet almost every acoustic need. Building from our expertise and product range that can deal with very low room modal issues to the very highs we use this approach to transform every aspect of the sound of your room, sculpting the sound and changing negative interference and modal issues into a positive and room sound changing experience.

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BIG PIG MEDIA LLP

WELCOME TO OUR PROMOTIONAL GUIDE TO ROOM ACOUSTICS IN PARTNERSHIP WITH GIK ACOUSTICS

Welcome to what we hope you will find an informative partnership promotion publication between HiFi Pig and room acoustics specialists GIK Acoustics.

As music lovers we aim to get the very best from our audio systems to maximise the enjoyment of our favourite performances, be that on vinyl, CD, streamed media… or whatever may come along in the future. In pursuit of audio-nirvana we often spend large sums of money on new electronics, audiophile media, higher resolution files, and other ancillary gear. All of these are important factors in reaching our stated goal, but one of the often overlooked aspects of the system and how it performs is arguably one of the most important. I'm talking, of course, about our listening spaces and how they interact with the soundwaves emanating from our loudspeakers.

To many, room acoustics and getting it right may seem like some kind of dark art – something only practiced by those running top-flight recording and mastering studios. However, with a few simple tools at their disposal and a bit of knowledge, anyone can make the most of the space they are working with.

The results of a well sorted room are undeniable and will help you get the very best from your audio components, whether they be modest in price or in the upper echelons off high-end audiophilia.

The long and the short of it is that room treatment is for everyone! And what's more, with GIK's Art Panels it can become part of your room's decoration.

In this informative and yet accessible guide, HiFi Pig has partnered with GIK acoustics to put together a promotional publication to help you understand and implement the basics of small room acoustics in your own home. In this guide GIK Acoustic's David Shevyn will introduce and explain the science behind room acoustics, demonstrate how you can, with a computer, a free program and a few simple tools, measure your room accurately, and finally implement effective measures to get the best from your space... and in turn from your valuable audio equipment.

Stuart Smith

SMALL ROOM ACOUSTICS

David Shevyn
GIK Acoustics

HOW SOUND WORKS IN YOUR ROOM

Normally when I give a lecture, I will begin with answering why we even need to treat a room at all. I demonstrate this by using two sound files of the same recording, using the same equipment, in the same room. The first sound file was recorded with the room empty (untreated) and the second was recorded using acoustic panels and bass traps. Even on a simple demonstration like this the audience is clearly able to hear the difference between the two recordings. In the first example where the room is untreated, vocals are echoey and the bass is muffled. Once room treatment has been applied the echo is removed from the vocals and the bass is cleaner and punchier. It's a simple demonstration but what it does illustrate is: the room you are in affects how you hear sound.

A LITTLE MATH GOES A LONG WAY

So, starting at the beginning we need to look at what the sound coming from your speakers actually is. What makes the music you are hearing? The notes in music all equate to a frequency. Technically a frequency is the number of soundwaves that pass a particular point in a fixed amount of time. We are all more familiar with Hertz, however, which is more specifically the number of waves that pass by in a second. So, 220 hertz is 220 cycles which is also known as the note A3. A soundwave isn't static, so we are also interested in the amplitude of the wave. This is how far the wave moves from its average position, i.e. how far the particles are displaced. The reason the amplitude is important is it comes across to the listener as the loudness, or volume. So, we have the musical note (the frequency) and then we have how loud you hear it (the amplitude).

As you can see in Figure 1: the top of a soundwave is referred to as a peak, which represents the loudest part of the soundwave. Then we have the low part of the wave, which is called a null, which is the quietest part of the soundwave.

A wavelength is measured by

Figure 1

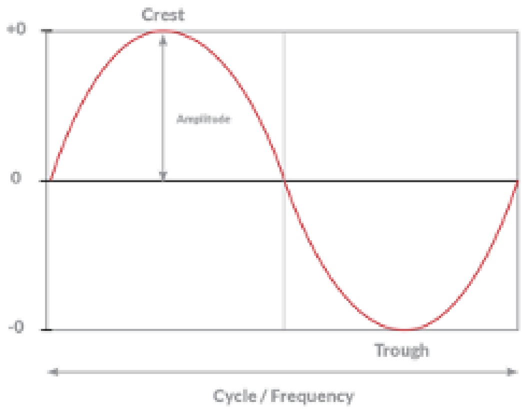
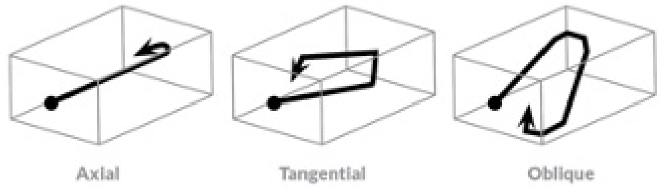


Figure 2



the distance between two peaks (or nulls).

You can work out a wavelength from the frequency and speed of sound (343m/s)

Wavelength = Wave Velocity divided by Frequency

So, if we take 220 hertz

It would be 343ms / 220 = 1.56

So, the wave length of 220 hertz is 156cm's.

A very low wave like 40 hertz would be

343 / 40 = 8.56 or 856cm's (8.56metres)

MODE DOWNLOAD

Now that you have a basic understanding of the physics of a soundwave, why is this relevant to your room? In basic terms: room modes are pre-existing resonances created by the room's dimensions. There's no way around it. The size and shape of your room dictate how soundwaves behave and react. Especially in the low end of 200hetrz and below

Room modes are activated when you play music and the soundwaves from your speakers hit the boundaries in the room. If the boundary is the same length as (or half or one quarter of) a soundwave, then they create what is known as standing waves. A standing wave occurs as a result of two different waves moving in two opposite directions. As they pass, they create interference which then make our nulls or peaks. This is because the boundaries are stopping the soundwave from fully decaying as they would if there were no boundaries and the energy remained in the room.

There are three types of modes in a room (Figure 2).

Axial modes are created between two opposite surfaces. Best to think of this in terms of length, width and height of the room

Tangential modes are created between four surfaces in the room - we most commonly see problems with this in square room

Oblique modes are created by six surfaces - less common but I have seen this type of mode created in Bay windows for example

Axial modes are the strongest and many times, the only ones that are considered. Tangential and oblique room modes have less impact per mode but are also more prevalent. A combination of tangential and oblique modes can cause just as many issues as axial modes can.

PEAKS AND NULLS AND WHAT TO DO

Room modes can cause both peaks and nulls (dips) in frequency response. When two or more waves meet and are in phase with each other at a specific frequency, you will have a peak in response. When they meet and are out of phase with each other, they cancel, and you end up with a dip or null in response.

Dealing with modes is accomplished by placing sound absorbers at a room boundary to minimize the reflections off it so there is nothing to combine or cancel. These boundaries include, but are not limited to: floor, ceiling, front wall, side walls, back wall, and all twelve corners. While treating corners is not a complete solution, placing treatments in corners offers the advantage of being at the end of 2 or even 3 of the room boundaries so there is a lot of benefit to treating that area. However, in some situations there are modal issues which require treatment of the rear wall or even the ceiling over your head that treating corners would not solve.

In conversation with GIK Acoustics President Glenn Kuras he mentioned the following analogy which can really help to understand how sound works in a room. He likened the room modes to as the shape making a ‘tone’ like a hollow block of wood. The shape and size dictate the sound and the lowest frequency it supports. Also, as you dampen the sound you are changing the frequency, but really the major effect is stopping it from ringing/sustaining. If you filled a small hollow box full of rags it would dampen/absorb the ringing and change the frequency, but more damping of the ringing then changing of the frequency which you want.

LOW END ISSUES

Moving on we are going to look at low end issues that you can (and will) experience in your room. We are starting here rather than say first reflection points or getting into the nitty gritty of diffusion and the art of scattering because this is where all your problems really do start - get to grips with the lowest common denominator and you are well on your way to getting a great sounding room and secondly because it is the hardest part of the room to get right. You need exactly the right products placed in the correct position to get the maximum absorption. It may be the trickiest part of the room to get right but it can be a fascinating journey learning about your room and achieving the desired results.

We've talked about how the shape and size of the room itself can affect how you hear the sound in the room and this is certainly one of the biggest influences on low end problems in a room, but it is not the only factor that creates low end issues. However, we will look in those in more detail later.

HOW DOES IT WORK?

What do we even mean by absorption? To understand it we have to take a step back and look at the Law of Conservation of Energy. This law basically states that energy can neither be destroyed nor created but only transformed or transferred from one form to another. So even though the energy can change from from a liquid to gas and back again, the amount of energy actually remains constant. The energy itself has not changed, only the way it is perceived.

When it comes to room acoustics this principle is being applied to sound. Here the motion of gas at a certain temperature is being caused to slow down via friction. This friction gives off heat thus conserving the total amount of energy and reducing the strength and intensity of the waves released. This is known as acoustic attenuation.

TYPES OF BASS TRAPS

Bass traps work by providing resistance to the sound waves bouncing around the room, so our bass traps need to provide a good resistance to these soundwaves often referred to as the gas flow resistivity of the material. In general, most bass traps fall into 2 different categories

Velocity Based Absorbers:

These are the most common type of absorbers and they're based on porous absorptive material. Because of this you will often see them referred to as porous bass traps. Basically, a trap like this will work simply by the conversion energy into heat, as previously explained. In general, this type of trap will absorb across a broadband of frequencies. The density, thickness and type of material will define which frequencies that they absorb and how much. Generally speaking this type of absorber is applied

GIK Acoustics Scopus Tuned Traps. They look very similar to the porous absorber but are constructed completely differently using pressure to tune the traps in a narrow bandwidth. They are designed to be as compact as possible.

when a broad range of frequency absorption is required (which is most small rooms).

Pressure Based Absorption:

This category actually includes a number of different types of absorbers, but the commonality is that they are pressure based. They are also known as resonant absorbers, tuned traps, pressure-based traps or narrow band traps. These are generally much more complicated and use factors such as depth, mass, size of air cavities and materials to deal with the absorption of specific very low frequencies. The key with pressure-based absorption is to understand the purpose and scope of when, how and where to use them. So, understanding your room and how it effects the sound is essential. These traps deal with peak pressure points and can be used in conjunction with broadband treatment.

LET'S GET TECHNICAL

The best way to understand how this the pressure-based absorption trap works is to break it down into two further categories; the tuned membrane bass trap and the Helmholtz Resonators.

Tuned Membrane Bass Trap – One of the advantages of these narrow band bass traps is that they are very compact. These work by creating a depth and sound pressure to a quarter wavelength of the frequency you are trying to absorb. Depth of the trap, the density and type of membrane used all effect the pressure within the trap to tune it to the right frequency.

Helmholtz Resonators – Most people are familiar with the concept here and many of you will have seen the demonstration of how this works in principle by blowing across the neck of a bottle. This works because of the way air works and that it has a natural 'springiness'. By this we mean that when you compress air its pressure increases and then it more or less goes back to its original form. The frequency of resonance is determined by the volume of air cavity, the length and diameter of the neck and this then absorbs the frequency you require.

HOW DO I KNOW WHICH BASS TRAPS TO USE?

Once you start looking into bass trapping and visit several different manufacturers websites, you will possibly be more lost than ever - which bass traps do I need? Although we have described just the two main types of bass trapping there are various different designs available using a combination of these theories. Products like wood faced diffusion panels actually use a form of resonation in their design caused by vibrations leading to frictionless losses.
The simplest answer to the above question is to take acoustic advice from a respected company and they can talk you through all the aspects of a room and how to treat it.

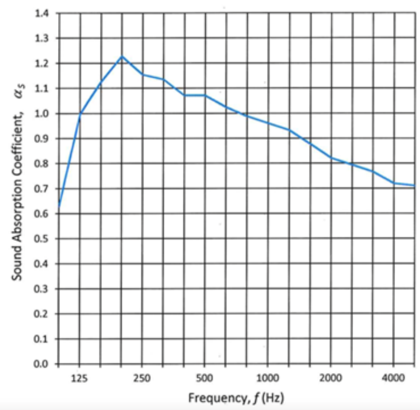
For now, let's jump forward a step and presume we do know which traps we want to buy, how to decide which one would be best for your room?

Reputable brands of acoustic treatments will have had their products tested in an independent laboratory and this is how we choose the correct treatment. Tests can come in different forms depending upon the type of product, where it was tested etc. but ultimately, we are interested in the "Absorption Co-efficient." This measures how well a material absorbs different frequencies. A perfect sound absorber would absorb 100% of a frequency and if it was to perfectly reflect a frequency then it would be 0%. These figures are sometimes written as a figure between 0.0 and 1.0

Figure 3

Random Incidence Sound Absorption Coefficient

Frequency [Hz]	α_S
100	0.63
125	1.00
160	1.13
200	1.23
250	1.15
315	1.14
400	1.07
500	1.07
630	1.02
800	0.99
1000	0.96
1250	0.93
1600	0.88
2000	0.82
2500	0.79
3150	0.77
4000	0.72
5000	0.71



The GIK Alpha panel is a porous broadband absorber with an 8mm wooden resonator in front.

The above data in Figure 3 is taken from an official ISO test conducted by University of Salford on GIK Acoustics 6A Alpha Panels. The Alpha panel is a porous broadband absorber with an 8mm wooden resonator in front. Official tests like seen above are conducted in a certified reverb room and need to cover 11 square meters, flat to the ground. This is known as an A Mount test. You will also often see J Mount tests. This is where the same panels are tested in the position they would be used (i.e. Manufacturers recommendation) rather than lying flat on the floor. All the numbers above 1 are 100% efficient. So, in the above example, the panel is 63% efficient at 100hz and 100% efficient at 125hz. This is a great broadband absorber that doesn't over deaden the highs and can go very low for a panel style absorber. Another factor to consider is the absorption area, the more surface area you have the more that it can absorb.

A simple demonstration for this is to consider 2 panels:
100mm thick absorbs down to 125Hz
200mm thick absorbs 63% at 100 hz and 100% at 125Hz

If you had a problem with 100hz it would not matter how many Number 1 panels you placed in the room. They would never absorb 100hz. However, we can calculate (based on your room dimensions) how many panels of number 2 we would need. Above 125hz the panels may then perform very similarly.

I always recommend that you work with companies that have these independent test results for two reasons
It is a certification to say that the product you are buying performs to a certain pre-defined criterion. Without this information these companies would not be able to advise you on how to deal with your room. It is essential to know how each product performs to be able to recommend the most effective acoustic solution for your needs.

WHERE DO I PUT THEM?

Again, this isn't a straight forward question or answer. It never seems to be, does it? This isn't us making the subject complex, instead our articles are designed to try and bring the complex physics of sound as it applies into our small listening rooms into something we can practically use in our own spaces.

Bass trap positioning can broadly be placed into three positions or categories

Corners - In an ideal scenario we would recommend placing bass traps floor to ceiling in all available corners but why is this? Firstly, we need to understand what we mean by a corner. A corner is anywhere where a wall meets a floor or a ceiling, so there are 12 corners in a room but only 4 tri corners. Bass builds up in corners as this is where two room modes meet. Why corners have such a build up of bass is more straight forward it is here that all room modes end. You can actually hear this in action yourself by just playing some music and going and listening in the corner. You will hear an increase in the low end.

Walls - Bass traps can also be placed on the walls and specifically the back walls. This is top deal with non-modal standing waves. These are basically standing waves created by your position within the room. As you move forwards and backwards though the room you will hear peaks and nulls of different frequencies so treating the back wall will prevent this. As the back wall is often the longest mode in the room then thick treatment is always required to deal with this.

Places of high pressure - When we are concerned about individual low frequencies, we are trying to absorb then it is essential they are placed in the correct position, specifically in the areas of high pressure for the frequency the trap is tuned to.

Figure 4 below shows an example of a 2 Channel (stereo) listening room set up. Note the floor to ceiling traps in the corners and the thick traps on the back wall behind the main listening position

SPEAKER POSITIONING

So far we have covered how sound works in a room, about room modes and bass trapping, before we look at how we test our room and

Figure 4

then how we resolve the issues raised by these tests we are going to look at speaker positioning and the importance of ensuring that you and your speakers are placed in the best possible position within your room. In acoustics, we term the problems created by Speaker positioning as SBIR.

WHAT IS SBIR?

Speaker Boundary Interference Response is the term to describe how the proximity of a speaker to a hard boundary (wall/ceiling/floor) will change the response, especially in the low end. In an ideal world all speakers would actually be played in a free field. A free field simply put is a space with no boundaries. In this scenario they would be no absorption, reflection, diffusion, the soundwaves would just decay as intended over the time and space required.

HOW DOES SBIR WORK AND WHY CAN IT BE A PROBLEM?

In a nutshell SBIR is primarily a case of distortion or interreference created by reflections from your speakers reflecting back from the wall and when they combine with the direct wave soundwaves it creates interference. As with all soundwaves, the problem occurs when the length of these reflections and the direct soundwaves correspond in wavelengths (and quartiles thereof). If they are half a wavelength out of phase, they cancel each other. Basically, the location of the speakers to the boundary has created another acoustical phenomenon known as COMB FILTERING. A comb filter is when a delayed version of itself (in this case a wavelength) interacts causing either constructive or deconstructive interference. A null is created when the signal is out of phase with each other by half a wavelength. If comb filtering was present when recording drums for example, for the placement of two microphones, the result would lead to a less punchy sounding recording.

So, to put all this together and if we recall what we discussed in my first section in this publication, every frequency has a different wavelength. If a delayed version of itself was to reach the listener at the same time we would get comb filtering. These delayed versions are created by reflections mirrored in the boundaries surrounding the loudspeaker. So, the position of the loudspeaker to the boundaries around them is going to change the interference we receive. The distance the loudspeakers are from these boundaries are going to measurable in both cm's and in which frequencies are affected. Move them and this interference will move to correspond with another soundwave that corresponds to the new distance.

As I said in my first article, nothing needs to be a problem in acoustics. Anything negative can be turned into a positive. Yes, we know that this interference exists but what we are interested in is how we make this interference work for us rather than against us. We are going to do this by moving the position of the speakers. Potentially nearer the wall, maybe further out but basically, we are going to MANIPULATE that interference until it is either very minimal or in a frequency range that we can easily deal with via normal acoustic treatment and techniques.

CASE STUDY– SBIR IN ACTION

In a break from the pure science I thought that the best way to really demonstrate SBIR in action was to walk you through a case study.

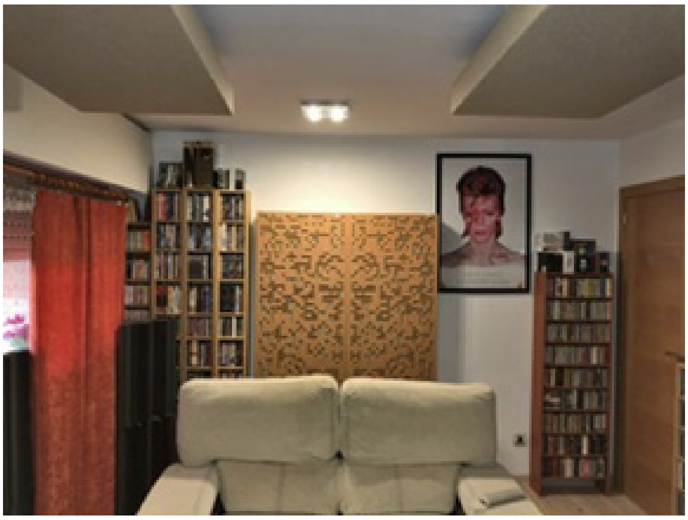
This is an actual GIK Acoustics customer who has been kind enough to allow us to use his experience for this article. As I go through this case study I will be introducing you to 3 different types of acoustical graphs. I will take a little time to talk you through each graph, what it means and what it is representing in this case study. Later we will delve into acoustical testing, how to do it and what it means in more detail.

The Room

This was a dedicated listening room that had already been advised on and treated by GIK Acoustic products. As with all scenarios, the room was treated within the confines of the space, budget and restrictions as agreed with our client. At the front of the room, each corner had floor to ceiling Tri Trap Bass Traps. 2 X GIK 244's was also on the front wall behind the speakers.

The ceiling was also treated with two clouds and (not in the picture) freestanding traps placed in the first reflection points. In the second picture we see the back of the room. The corners here are inaccessible to treatment but the back wall has 2 X Alpha 2D panels on them to both absorb and diffuse.

Here's a couple of images of the room to give you a better idea of what we had to work with and how that might translate to your own room.



On the whole our client was super happy with his room. However, he was experiencing one large problem. A null at 51hz. He ran some REW EQ Wizard tests (a freely available room testing software we highly recommend) and presented us with the following three graphs. I will explain each graph and its meaning as we go.

Figure 6 is a Sound Pressure Graph. It measures the loudness of each frequency. What we try to achieve is to get the difference between each frequency as level with each other as possible. As you can imagine, if 60 hz is much louder than 70hz, the 60hz would drown out the 70hz and thus the music would not be reproduced as intended. On the left-hand side, we have the db's (the loudness). Across the bottom we have the frequency recorded. The problem here is the large null (or lack of loudness) of 51hz.
However, the graph only tells us the loudness

Figure 6

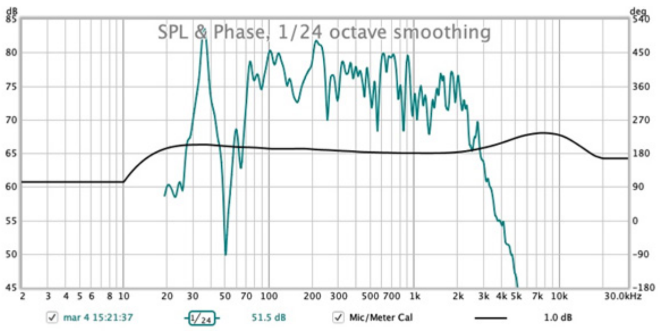
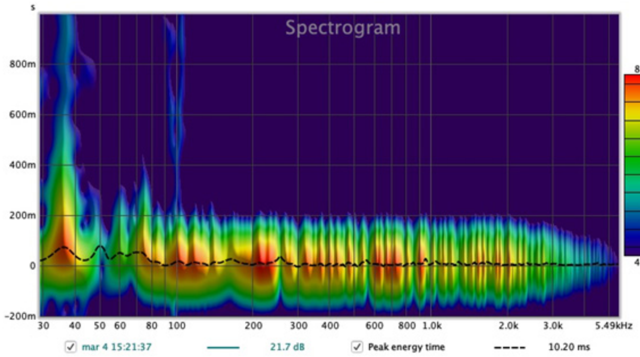


Figure 7



of each domain. We also want to know what is going on with regards to how long each of those frequencies remain in the room. For our case study we are just going to concentrate on the Spectrogram Graph (Figure 7).

You may have seen in the past a waterfall graph representing decay time. This is similar, it is that waterfall viewed from above. Along the left-hand side, we have how long each frequency rings out in the room in m/s (or the way we look at it, how long it takes to decay). The bottom is again the frequency. As an added dimension to this graph is the colour coding. That is pressure (in db's like our SPL). Red is the most intense, dark blue the least.

In this scenario we have a ringing at about 37hz for over 800ms and 51hz is nonexistent (this is our comb filtering scenario).

What are we looking for with a spectrogram? Every room is different, but we are looking to reduce the ringing of each frequency to between 140ms and 240ms. As you can see above 80hz we have achieved that in this room. Is it the very low end we are struggling with.

So, what did these graphs tell us? In simple terms the large null then reproduced with no decay time on the spectrogram shows comb filtering in action. There are other graphs and information that we also have provided but all the indications pointed towards it being a problem with SBIR.

Tests Following Moving the Speakers Closer to the Wall.

This time the SPL graph (Figure 8 below) has both results on it for comparison. As we moved

Figure 8

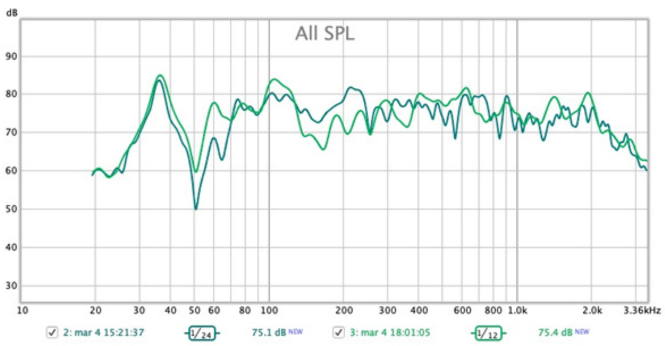
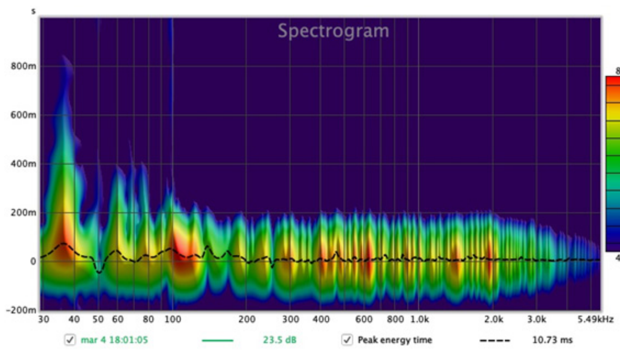


Figure 9



the speakers closer to the wall the results started to change. We had changed the distance to the boundaries therefore the frequency response also changed. It took 10 dbs off that 51hz null.

What about our Spectrogram (Figure 9)? We see our 37hz problem has been reduced but still exists and that null is still there. Surely, we could do better? Well in this case we could, and this is why acoustics can at times be quite complex and confusing and why we would always recommend getting the advice from a professional where you can. In this case we asked our client to remove the acoustic panels from behind the speakers.

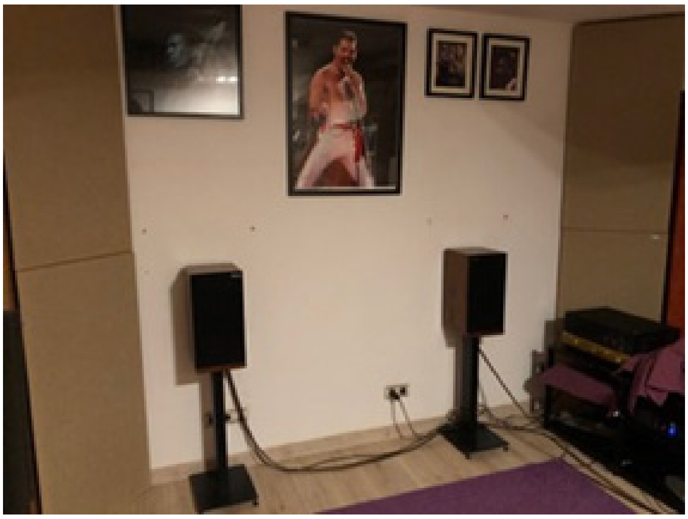
There are so many factors that come into play with SBIR, the loudspeaker and how they have been manufactured means that sometimes the only way to tame SBIR is with a 100mm panel but sometimes having a panel so close to absorption will make it worse. In this case we asked our client to remove the acoustic panels from behind the speakers and place them in the rear corners.

So, this time I have left the previous reading on and compared it to our new reading. Look at the difference. The peak at 37 hz has been reduced by 15db's, the null at 51hz by about 12db. Overall the room is now looking and sounding pretty good for the treatment it has in it.

Relating this to the Spectrogram –

The difference is even clearer to see. That 37hz decay time has literally been cut in half. This room is looking great and this has been achieved by a combination of excellent positioning and acoustic treatment.

Speaker Boundary Interference Response need not ever be a problem, it can certainly be made to work for you rather than against you. What I personally always recommend, and it is indeed one of the most important things you can do when setting up your room is spend the time to position the speakers to get the best results. If you do that first, it makes treating the rest of the room so much easier. Just moving the speakers bit by bit and testing them is the best way to start and from there you can start to build up a picture of how the sound is working in your room, with your boundaries and modes in place.



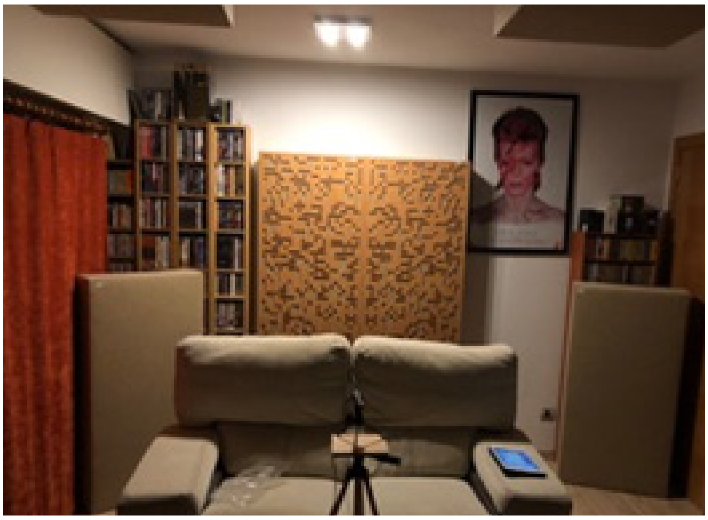


Figure 10

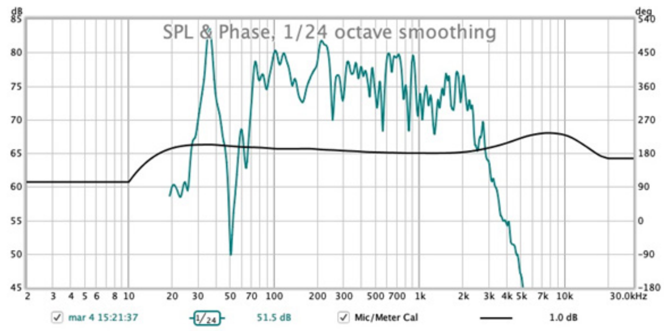
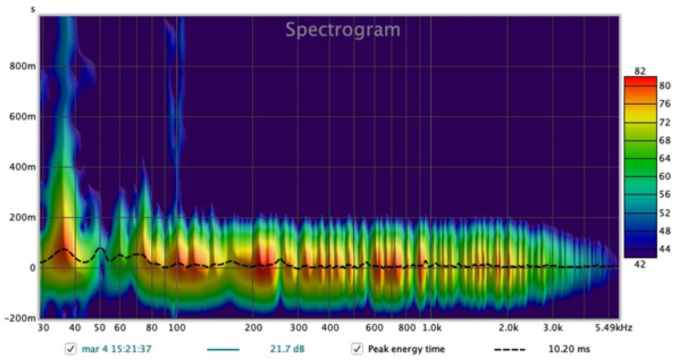
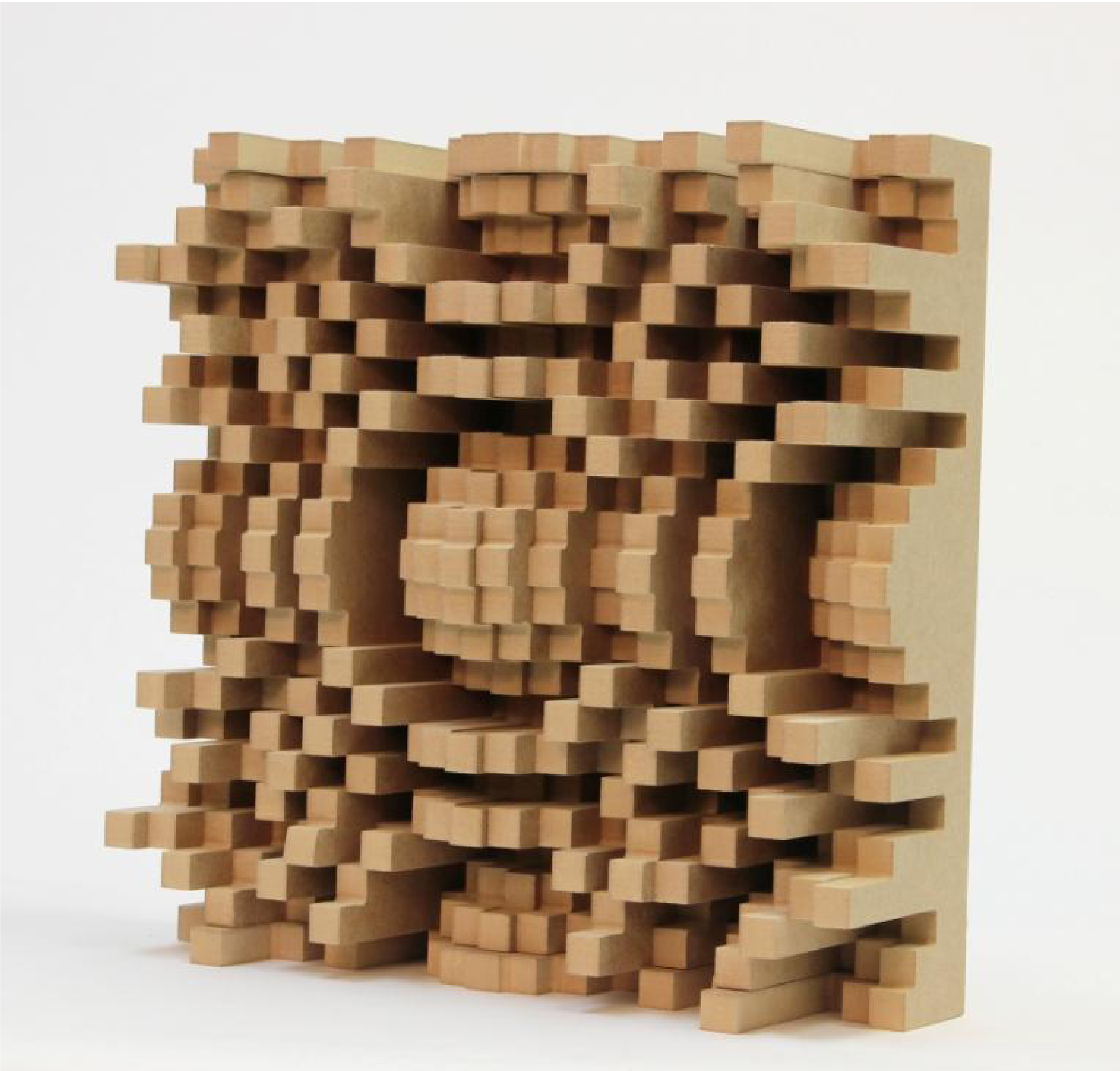


Figure 11





GIK Acoustics Gotham N23 Skyline Diffusor

TESTING TESTING TESTING

More than ever now, when it comes to discussing acoustics on forums, applying acoustics, or for example looking where to place speakers in a tricky shaped room you will hear people ask for your 'test results'. It has almost become a caveat to all recommendations. If you are considering two or three set ups in a room for example, almost all acousticians now will give an opinion and it will be followed up by I would test this first. So why testing, what is it and how do we do this for ourselves?

SO WHY TESTING?

One of the largest issues within the Pro Audio community is that it appears that everybody seems to have a different opinion of how to fix an issue, how to build traps, which traps to use and where to place them. In fact, on almost every acoustics theory or solution, someone, somewhere will have written the opposite view and in both cases, it almost certainly will be written with authority that they are right. So why is this? Two reasons instantly spring to mind, and both relate to physics. The first is that due to physics, and how sound interreacts with everything that is in a room, although certain acoustic models can be used it is extremely difficult to accurately predicts how sound is going to react in a room so acoustic testing actually shows us exactly what is happening in that room in a given moment in time. The second reason is that physics is an extremely difficult topic to master and it yet again another case of a little knowledge can be dangerous.

What is acoustic testing?

Acoustic testing in our small room scenario is a way to measure what is happening to sound that is introduced into your room once it has left the speakers. By the use of multi directional microphones and acoustical testing software which both provides frequency and impulse responses and measures that response we are able to measure aspects such as distortion, phases, decay, frequency loudness (level) and reflections amongst other measurements (data). Basically, we are able to build up a picture of how sound is reacting to the room, the objects you have inside that room at a given point in the room and at a given point in time. Once this information has been gathered, we can start to see exactly where your problems are in the room and start to consider our strategy to deal with it. What is even better, is this process is so simple and quick that we can continue to use the software to measure, makes adjustments, test hypothesis' and even EQ individual frequencies if needed. Basically, this simple process takes the guesswork out of acoustic consultation.

HOW TO TEST

In the rest of this article I am going to talk you through how we test. From which microphone and software to use, to setting up and running the program. Feel free to contact me via HiFi Pig if you have any particular questions about this process and we can perhaps run a FAQ section on our next article.

THE SOFTWARE

We recommend the use of Room EQ Wizard for your acoustical room testing.

Why?

This excellent piece of software is free to use (they ask only for donations if you wish).

It is pretty much glitch free, really easy to run and can be used for both the room testing and EQing your room later if you wish

They have an excellent online forum and community who are more than happy to answer any questions or queries about the software that you might have.

Download it for free (or make a donation) at https://www.roomeqwizard.com

THE MICROPHONE

There are two approaches with regards to microphones and that is either to connect a mic via your audio interface or what is most commonly used these days due to simplicity (and the method that I am going to use in these notes) are measurement microphones with a USB connection.

Firstly, what type of Mic is that it you need? You need an omni directional small diaphragmen condenser mic. It is unlikely that you will already have one of these in your arsenal unless it came with room adjustment software. Basically, it is a measurement microphone.

We recommend the Mini DSP UMIK 1 / 2 as the best and most usable USB Measurement Mic in the market currently. It is accurate, easy to use and each mic comes with its own calibration file. Currently retail approximately 100 Euro.

GIK ACOUSTICS' SIMPLE STEP BY STEP GUIDE TO TESTING YOUR ROOM

GIK Acoustics Simple Step by Step Guide to Testing Your Room.

WHAT DO YOU NEED?

A measurement Mic (We are using the Mini DSP UMIK 1)

Measurement Software. We recommend the Free Room EQ Wizard Software. Download at **https://www.roomeqwizard.com**

A Computer

A connection to your monitors

WHAT NOW?

1. Download the Free Software for your computer operating system and install (I am using a Macbook Pro in this guide).

2. Connect your Microphone to your computer.

Once connected the small blue light will display

3. Open the Room Eq Wizard Software. If you have a calibrated mic like the Mini DSP UMIK 1 it should recognize it and ask if you have a calibration file. Figure A.

4. If you have the calibration file, it will ask you to select it.

5. If you do not have the calibration file, then go to MiniDSP.com and on the Microphone page it will ask you to enter your serial number. It will then allow you to download two calibration files specific to your microphone. For the purpose of this exercise we want the one named after the serial number. Do not use the 90 degree Axis one here.

6. Next click the measure icon in the top left-hand corner of the screen. FIgure B.

7. You should now be on the Make A Measurement page, which looks like Figure C.

8. Check the settings. Type should be SPL and method Sweep. You also have your first opportunity to name the file here.

9. Check the Start and End Frequency. Set the Start Frequency to 0 and the end Frequency to 20,000hz

10. Setting Length should read 256K

11. Sample rate should be 48khz and set the delay to 0. The delay setting is in seconds and can be used if you wish to move from the mac to a different position for the start of the test. It is a delay from when you press test until the test begins

12. Input – This should be your microphone.

13. Positioning of the microphone. For this first test we are ideally going to place the microphone in the listening position. So, remove your chair (ideally well out of the way, I have seen many confusing reflections show on results from high backed reflective chairs) . The Microphone needs to be ideally at ear height and pointing towards the ceiling in your normal sweet spot. Figure D.

14. We are now ready to check levels. Press the Check Levels button

15. You should hear some white noise. It will now tell you the result. Figure E

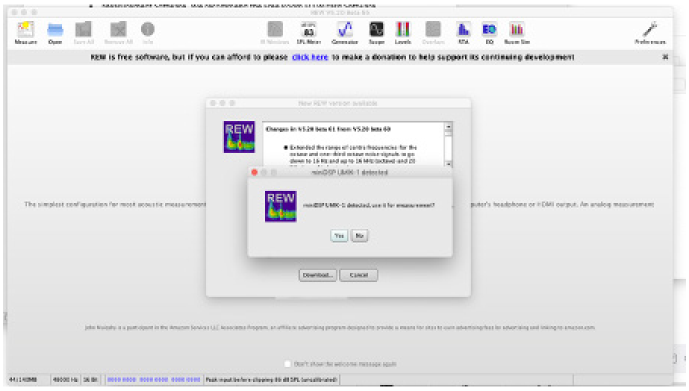


Figure A

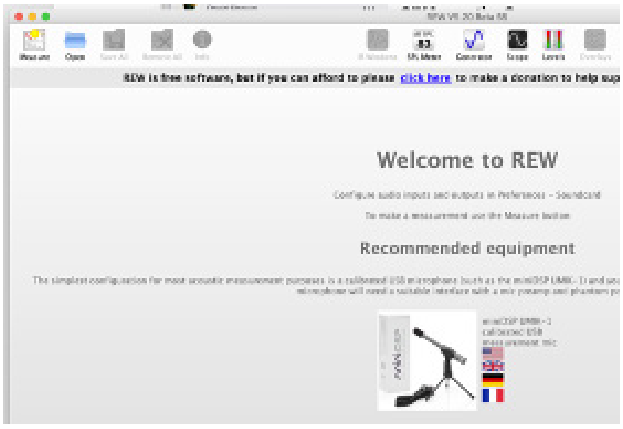


Figure B

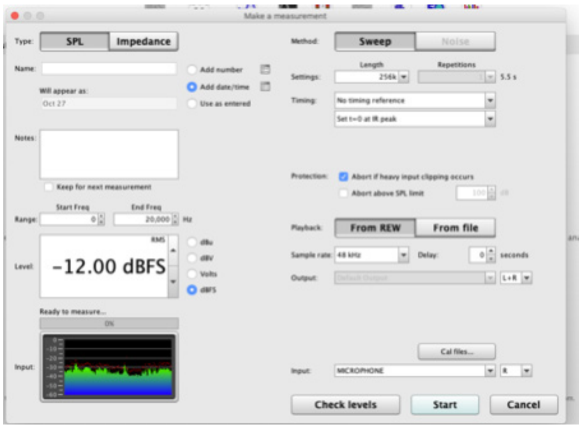


Figure C

Figure D

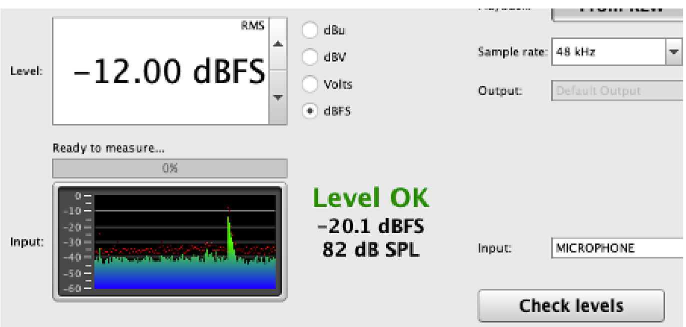


Figure E

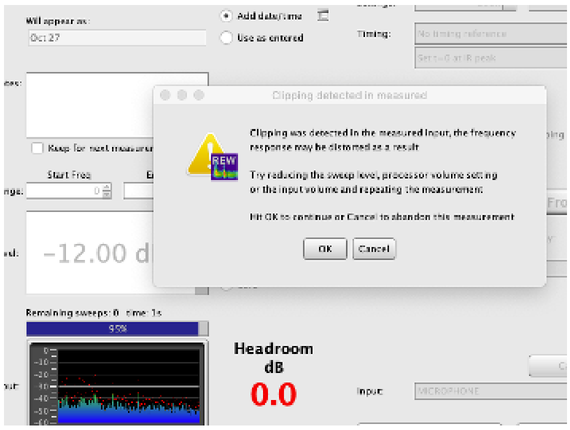


Figure F

Figure G

16. My level is set to OK which means I can begin. Note that it is about 82db's which is normal listening levels

17. Before we actually being the testing make sure that you have the room as silent as possible. Do not talk or make a sound, if there is noise from outside wait for it to stop before you commence. Also check that such items as air conditioners have been turned off before you begin

18. Now we can begin. Press the start button. Please note, it is always wise to cover and protect your ears whilst the sweep is taking place.

19. If the settings are still not right, you will get a warning as per Figure F. As you can see, I left no headroom so clipping occurred. Turn down the volume and recheck your levels before pressing Start again

20. In this example I am using testing the stereo position, but we recommend you repeat these steps for the left monitor and the right monitor separately as well.

21. In a few simple seconds you should see the screenshown in Figure G.

22. If you do, you have now successfully captured a sweep of your room

In the next section we are going to talk about what these results mean and how to interpret what that means for your room.

HOW TO INTERPRET ROOM EQ WIZARD'S RESULTS

So, assuming you have managed to follow the instructions of our last article and have successfully got a reading of your room. Now how to interpret it?

At first glance it seems super confusing - there is a lot of information and how does it all relate to each other. In this article I wanted to talk you through just some of the main functions of REW, the graphs and generally what they mean. In this basic introduction we hope to get you started but in reality, when an experienced acoustics advisor is using this information, they will be looking at a lot of different factors to help interpret what the graphs are actually saying. They will also:

Cross reference the graphs. How they inter react with each other tells us a lot about how the sound is interacting within the room.

Look at the dimensions of the room and make room calculations based on those dimensions, possibly in multiple points of the room, if there are different areas or heights.

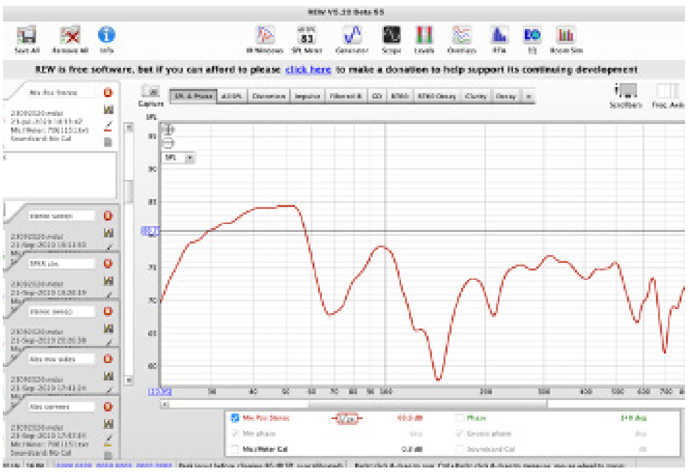
Consider the frequency range of the speakers and whether any subs are being used. Also consider what we know about how different monitors react to boundaries.

Consider the materials, thickness and shape of the room and how that will affect the way that sound works in that room.

All of the environmental, external and life pressure restrictions that will affect how we interpret the sound and how the sound reacts to the physical dimensions it enters.

GUIDE TO THE GRAPHS WE ARE GOING TO STUDY

Hopefully you should have a screen similar to the below. It will be automatically set to the First tab SPL & Phase.



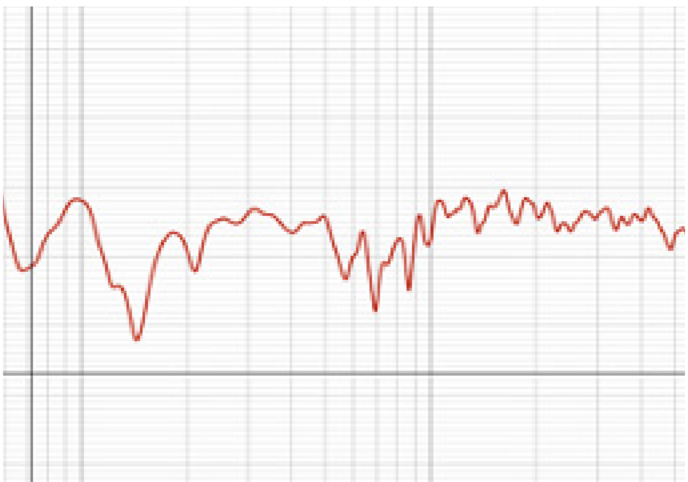
In this article we going to be looking at this tab (and the corresponding All SPL tab and Impulse Graph tab. In Part 2 will then go on to look at the RT60 tab and the waterfall and Spectrogram tabs (not shown here) and some common issues you may come across.

THE SPL (SOUND PRESSURE LEVEL) GRAPH

Our first graph is one that I sure am a good number of you are already familiar with and will have seen this in some sort of context, if for no other reason than produced by manufacturers to show how their monitors perform.

But what exactly is Sound Pressure, and what are we measuring? Sound pressure itself was defined as the difference between the actual pressure of a sound wave and the average pressure of a sound wave. What this is often demonstrated as is as decibels, literally how loud each sound wave is, or put another way, the intensity of each sound level but as the intensity is very difficult to actually measure then it is the sound pressure which we record via sound pressure levels (or your REW Software in this case). Sound pressure is approximately equal to sound intensity. Decibels is the scale we use to measure this pressure (or intensity), the more pressure, the louder it is and higher up the Db scale it will be. A decibel is basically one tenth of a bel and this scale works closely to how we hear and interpret the loudness of sounds.

So, let's take a look at a graph.



X Axis is the sound pressure levels measured in decibels
Y Axis is the sound frequency that is being measured in Hertz.

Settings – Go to graph and choose 1/24 smoothing

Hopefully you will have measured and captured your test at normal listening levels which would be about 75hz. Your normal listening level should not exceed 85hz for prolonged periods of time as this will cause irreversible damage to your hearing in the long term.

The first aspect to note is where the results will begin. It will depend on your monitors, but if you are using some small bookshelf monitors the readings may not really kick in until about 50hz, it will depend on the speakers being used. In our example above it is clear these speakers work from around 28hz.

HOW TO INTERPRET THESE RESULTS

The line on the graph is tracing the sound level of each frequency as the sweep was played and the results recorded in your room. For example, 50hz is 84.2 dbs whereas 70hz is 68 dbs. Each frequency in the scale that your monitor produces sound has been recorded this way.

This means that note G1 (50hz) is 16 dbs louder than note C#2 (70hz). Imagine how that would sound if you were mixing a track, you literally would hear G1 super loud and C#2 not at all.

WHAT IS IT WE ARE LOOKING FOR?

In an ideal world the difference between any of the frequencies would not exceed 10db's. If it does, these are the frequencies that we are concentrating on providing additional treatment for. However, in the real word almost the best you will see is a 6db difference. I have only seen maybe 3 better professional studios than that in my entire career, so you do have to be realistic with what we are aiming for.

For the graph above what key areas would give me cause for concern?

The entire low end would need cleaning up. The movement from 50hz to 180hz has fluctuation of plus / minus 25db. The mids 643 to 700hz has over a 10db fluctuation and the highs certainly could do with some taming.

Our graph shows an untreated room, but you can keep going with this process for various different reasons, to find the best speaker position, once treated to fine tune etc.

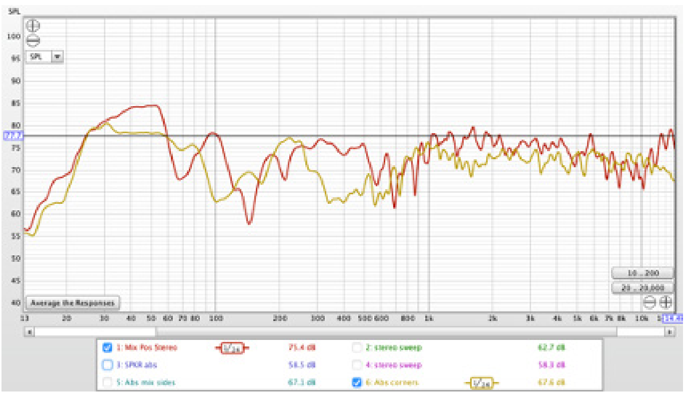
THE ALL SPL GRAPH

It is worth briefly mentioning this graph, as basically it is a comparison tool.

Every test that you conduct and save to the same REW file can then be selected and compared against each other. If you are trying to find the best speaker position for example this is great information. There are many different situations where this would be useful:

- If you are trying to isolate a certain issue then, taking a reading of the left monitor, then the right monitor and then stereo can tell us a lot more about the room and where that issue is being generated from
- To compare the effectiveness of different acoustic treatments in different positions
- To find the sweet spot in the room
- To find the best monitor position.

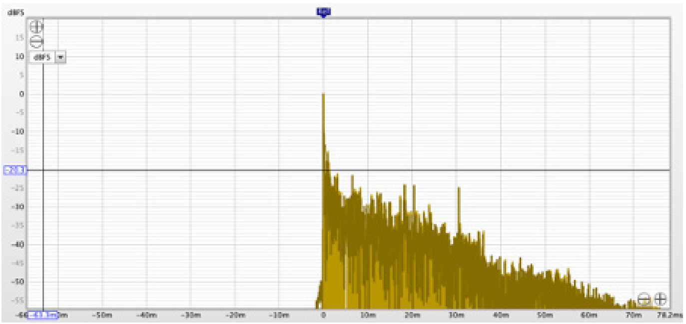
In our sample graph below, we are comparing our original mix position reading with a reading once some Alpha panels had been placed in the front corners. Look at the difference that they have made in the low end. At 50hz it has reduced the decibels by 6.5 dbs and at 70hz it has raised it by 6.3 dbs. This means our previous 16dbs has now been reduced to only about 2dbs!!



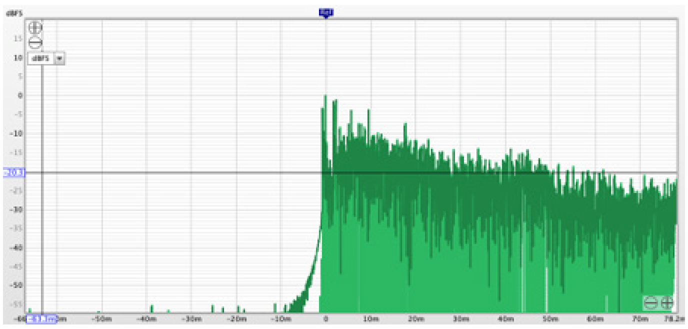
The Impulse graph is our first example of the large capabilities of the Room EQ Wizard software and what it is able to record and produce but in the context of this article and for the purposes of a means as an introduction to the software we are going to concentrate purely on the Impulse Response Envelope or in simple terms the reflections within the room.

What is an impulse response graph? It is a graph that plots what happens if you played a single very loud noise in a room. What we are interested in is what happens to that sound once it is released into the room and what it then says about how that room is affecting the way we hear music. The original impulse response tests were actually generated to plot phase inaccuracy (and a lot of what REW does now enables you still to do this) particularly in the design and manufacturing of loudspeakers. In our context we are using it to plot the acoustic characteristics of our room.

To understand this best, I have prepared the two following graphs. Graph 1 is in a fully treated room.



Graph 2 is in a completely untreated room.



X Axis is the sound pressure levels measured in decibels; in this case we have recorded the drop of sound level from the initial sound played into the room.

Y Axis is the length of time in milliseconds that the microphone recorded the reflected response.

What are we expecting this graph to tell us?

As the graph is recording the way that the sound has reacted within the room at the point it meets our microphone this graph is telling us about the reflections that it receives. These are most recognizable as echoes in a room. A sound has been released into the room, it has hit every surface it comes back into contact with, bounces around the room and eventually comes back to our microphone. How quickly and at what loudness (pressure) that reflection is received makes a difference to how we as humans comprehend what we are hearing.

We can hear it, we know it is there and we can deal with it by treating our flat, parallel surfaces but what about the sound that reaches our ears in that window that has dropped by less than 20dbs and reached our ears within 20ms'? (20ms is a rough guide for smaller rooms, this can move up to 30ms dependent upon the size of your room) This is the problem and the curse of the mixing engineer. Our brains cannot compute this as a separate sound from the source and therefore it picks up mixed signals, it may think it is hearing the left speaker when it is hearing the right and it can create a situation known as smearing, where the reflected sound reaches our ears so quickly after the original source sound that instead of a crisp start and finish to the sound or frequency you get a smeared confused sound instead.

How do we interpret the graphs?

· Every line on the graph is reflected sound. I have drawn a line across the graph at -20db's. On the second untreated room you can see that there are many, many lines above this. That is all the reflections bouncing around the room.

· In Graph 1 where most of our reflection points have already been treated you can see that there are very few lines above this threshold and in general the graph reduces as many reflections have already been dealt with and it is just the few bouncing around the room we are seeing.

Graph 1 does have some troublesome lines. What are they? What is great about sound is that it is linear which means of course we can do a calculation to work out, what distance the reflected sound has travelled before our mic has picked it up. If like in example 1 the lines are very close, it is often something super close to the source and the mic, the biggest culprit here is the desk which is difficult to treat and why we would always recommend using the smallest desk possible to be able to achieve your tasks. It can be other problems, I once had a situation, in a fully treated room, with a nasty 150Hz problem that I could simply not pinpoint. It wasn't until I used this graph and some calculations and then eventually received a photograph of the room that I was able to identify the problem had been a very high-backed chair next to the microphone all along!

How to calculate where the reflection may have been generated:

- Check the graph to ascertain what the ms of your problematic reflection you want to calculate the distance of is. Let's say it is 2.5ms.
- Sound travels at a speed of 343 metres a second
- Or 343mm a millisecond

· Times by 2.5 and that sound would have travelled 86 cm's

- Then it is a matter of working out what could have travelled 86cm's from the source to a reflection to our microphone.

Whilst looking at the graphs and making your own interpretations from them is fun and useful and fun exercise, GIK offer a full evaluation service whereby you can carry out the tests as described herein and send the graphs off to an expert acoustician who will suggest the best way to treat your room to maximum effect and within your budget. You can find out more here along with other useful free tools

PART 2 COMING IN 2022