

Hello everyone.

My audio career began like many others, mixing live sound for local bands and musical theater. Over time I worked up to bigger acts and bigger venues.

From there I moved to live recording with a remote truck I built, and then into broadcast. Seeking more consistent employment, I shifted to the hardware side, eventually working for the legendary Deane Jensen.

I then started Black Audio Devices, selling mic and boom stand parts and other things I invented.

Finally, I moved into post production sound engineering, as both a staff and freelance engineer. I've now spent over 25 years working for such companies as Skywalker Sound, DreamWorks, the Academy of Motion Picture Arts and Sciences, and others.

During that time, the acoustical problems often ended up in my lap. But the information available to solve them was basically hearsay, guesses, and "secret mystical knowledge". This made fixing problems *effectively*, very difficult, and designing good acoustical spaces a hit or miss affair.

I instinctively felt that acoustics was science, with no secrets or obscure knowledge, and that it was possible to create high performance acoustical spaces consistently. I ended up reading and rereading many books by such authorities as Everest, Salter, Long, Toole, and others.

This led to developing my own technique for designing rooms, that has proven to be consistently successful. I have created critical listening spaces, provided solutions for existing room's problems, and consulted, for the likes of Sony Pictures, Technicolor, iHeart Radio, and Simpsons creator Matt Groening, as well as post production legends like rerecording mixer Paul Massey, and music mixer Bruce Botnick.

I have been a member of the Audio Engineering Society for over 33 years, the Academy of Motion Picture Arts and Sciences for over 20 years, serving on its Theater Standards Committee, and have been published in Mix Magazine, Recording Magazine, the Cinema Audio Society Quarterly, and the Motion Picture Editors Guild Magazine.

In 2009, I officially hung out my shingle as MediaRooms Technology LLC.

If you'd like to contact me, my email address is in the bottom left corner.



Now, it has been written that money is the sum of all blessings and the root of all evil. We could say the same of low frequencies.



The low frequency region is where the bulk of our perception of sonic quality resides, so when things are going well there, we are blessed with a pleasing, accurate listening experience.



But when there's trouble in these low places, we are tormented with the mess of an uncomfortable, fatiguing, and confusing listening ordeal.

And if we try to figure out what's causing these problems and fix the issues, we can find ourselves stymied and unsuccessful, running in circles, and chasing our tails.

This is because our common, traditional method of diagnosing these types of issues leaves out crucial data, with far more information missing, than what actually <u>is</u> revealed to us.

And hand-in-hand with this, our uninformed attempts to fix these problems frequently employ the wrong tool for the job. In the end, we just take blind pot shots in the dark, hoping to stumble on a lucky break.

At best, an ineffective approach.



So the low frequency region carries a lot of responsibility for the accuracy and quality of our listening experience, and as a result, the quality of our product - whether it's music, a film or TV show, or a podcast.

It can make it... or break it.



To get a handle on this and be able to control the low frequency performance of a room effectively, the most important thing to understand is that sound has two distinct ways of propagating.

When we think of sound transmission, we pretty much always think of the example of sound traveling like little projectiles being shot around the room.



We often see this illustrated as the peashooter method, where a stream of peas is aimed at a wall, or perhaps some unsuspecting victim. This type of sound propagation is particle motion.



They bounce and reflect off the wall, obeying the "angle of incidence equals the angle of reflection" law of physics we are all so familiar with. This is known in the scientific realm as a "specular reflection".

But sound also travels by pressure. This differs from particle motion in that some acoustical energy can travel without the air particles having to move hardly at all. You could say that instead of running around bumping into each other like a pinball machine, it's more like a crowd surge at a soccer match.

This makes things more complicated, but if we <u>really</u> want to be effective in our efforts...



...and not just throw a Band-Aid at problems, we <u>have</u> to dive in to understand how to deal with this increased, more baffling, complexity. But my goal is to dispel the confusion.



Pressure is the pushing force that particles exert on the things surrounding them.

And it increases, as those surrounding materials increasingly resist the movement of the air molecules. This causes the pressure to go up, and the particle velocity to go down.

So as sound approaches a solid object, like a room's surfaces, things change.



We can get a better idea, with an executive toy called Balance Balls. This is more formally known as a Newton's Cradle, demonstrating the laws of conservation of energy and momentum. Five balls are suspended in a row, and the ball at one end is lifted and released. The balls in the middle don't move, while the one at the other end flies out, illustrating that the energy, as pressure, still travels, even without the "particle motion" of the middle balls.

And it also reminds us that energy is neither created nor destroyed, except under some rather extraordinary circumstances. So there is no such thing as "the energy just goes away." It goes somewhere, it just may slip off our radar in the process.

So in a room, as sound approaches something that is less likely to move than the air, the particle velocity goes down, and the pressure the particles exert goes up.

And the more immobile that object is, the stronger this effect. In any room, this is pretty much always the walls, ceiling, and the floor. And the airspace closest to these surfaces, where the pressure is highest, is called *the pressure zone*. What a surprise!



So this makes things more complex than the common, pea shooter model of sound transmission would have us believe. If all this seems a bit confusing, you're in for a treat - there's even more to it.



First, these two propagation methods act on different frequency ranges.

Generally speaking, higher frequencies travel by particle motion, and low frequencies travel by pressure, and there is a frequency range where it progresses from one to the other. This transition zone occurs around 300 Hz., and is called the Schroeder frequency.

Remember this little tidbit.

Oh, and I wouldn't taunt the sumo wrestlers...



Secondly, to **<u>effectively</u>** treat these two different types of propagation, we need two **completely different types** of absorption.

That means there is no "magic bullet" that absorbs all sound at all frequencies. We can't just glue some sort of squishy material on the wall and have <u>all</u> the sound be absorbed equally and completely across the entire audio spectrum.

For all practical purposes, that doesn't exist. It would make life easier if it did.



So, if we want to absorb <u>particle motion sound</u>, which is pretty much everything above 300 Hz. or so, we need a material that reduces particle motion with friction. This changes the energy of the particle motion to heat. But don't bother trying to take the chill off a room with this. It doesn't make <u>that</u> much heat.



So this would be things like the ever popular Owens Corning 703 semi rigid fiberglass board, shown here on the left.



It also includes absorptive materials like Ultra Touch recycled denim insulation,



...and fluffy fiberglass itch and scratch insulation, as well as rock wool, those infamous sculptured foam products that I enjoy disparaging, and other such materials.

These are all called resistive absorbers. Most of these materials are easy to get and build into high performance absorbers that are also visually pleasing.

But it's also <u>very important</u> to understand that the <u>thickness</u> of resistive absorbers like these determines how low a frequency they absorb to. So for example, a one inch thick resistive material absorbs reasonably well to just below 1 kHz., while 2 inch will absorb to around 300 to 400 Hz.

Unfortunately, this nugget has yet to be incorporated into common audio knowledge.



But the sad fact is, absorption is the treatment used most often, to treat <u>any</u> acoustical problem. We just throw some absorption at it, and hope that makes things better. But that almost never happens, making this a classic "Band Aid" repair.



In fact, if we take acoustical measurements in a room with a lot of absorption, perhaps too much so, we can often tell just how thick the insulation is without actually seeing it.

Here we have two response graphs of one room, covering 10 Hz. To 24 kHz. The upper trace is the direct sound from the speaker, while the lower trace is the exact same sound, but 120 ms. later.



At 120 ms., the room's acoustical treatments have kicked in. In this case, we see a "knee" in the response at around 1 kHz.

Since we already know that 1 inch thick resistive absorbers start losing their effectiveness at around this frequency, we can be pretty confident that this room has 1 inch thick absorption.

Likewise, if we get this "knee" at 300 to 400 Hz., the room would have 2 inch thick absorption.

Armed with graphs like this, your builder can no longer fool you by installing one thickness, and claiming it to be another.

This also illustrates a very important point. When you add a lot of absorption to a room, for example, and you hear a change, you can be lured into thinking this change means the sound is better. This graph shows that the sound is indeed different, but now the room is acoustically inaccurate. It's a big stretch to call this "better", when it <u>does not</u> accurately reflect what is actually in your recording. This will make it sound even more different as it's played in different rooms. And if you do "audio-for-hire", this could cause difficulties in trying to advance your career.



On the other hand, if we want to absorb **pressure-propagated sound**, we need to use resonant devices that respond to pressure fluctuations.

Again, energy is neither created nor destroyed, only transformed from one form to another. So we would use something like this Helmholtz resonator, or flat panel absorbers, perforated panel absorbers, or similar things. They turn the pressure fluctuations into motion, which is then dissipated by friction and heat.

But there's another important question - when do we use resistive absorbers, and when do we use resonant absorbers?



I'm glad you asked. This is where that bit about the 300 Hz. transition range comes into play.



Armed with these two types of absorption, we can essentially "tune" a room's entire spectrum of sound. To absorb sound below 300 Hz. or so, we use an absorber that works on pressure. Above 300 Hz., we employ an absorber that works on particle motion.

That would be resonant absorbers below, and resistive absorbers above. Simple.

But there's one more obscure consideration that's easy to miss - we also need to **place** them where they'll be the most effective.

Pressure absorbers need to be placed where the pressure is highest, and resistive absorbers go where the particle motion is highest. OK, common sense.

For pressure absorbers, that would be very close to the room surfaces, in the pressure zone we spoke of. And in fact, for the Helmholtz resonator shown here, it works best when the port is <u>facing the wall</u>, and within an inch or two of it. That puts the port as deep as possible into the pressure zone, while not restricting air flow around the port.

Meanwhile, resistive absorbers should be placed where particle motion is highest - <u>away</u> from the room surfaces.



This means all these eons that we've been gluing resistive absorptive panels <u>on</u> the walls and ceilings, we haven't been putting them where they do their best work! This doesn't mean they **won't** work, it just means they won't perform the best.

And for low frequency absorption, should we <u>really</u> be using <u>resistive</u> devices made of foam? Again, it's the wrong material for the job. That's why if you install them, and make a <u>truly honest</u> appraisal of it, you'll find the results are just disappointing.

Another Siren call onto the rocks and shoals of an acoustically inaccurate room.



So the next step in understanding low frequencies, and dealing with them effectively, is understanding room resonances. These are also known as room modes, eigenmodes, standing waves and so on. I prefer calling them resonances, because I feel that best describes what they are.



Every enclosed space of any shape will have resonances. You <u>cannot</u> get rid of them, and they occur throughout the entire audio spectrum.

Here we have a diagram of some low frequency resonances in a rectangular room, and in an irregular room of the same volume. This illustrates that we can <u>never</u> get rid of resonances with non-parallel walls, but we can most certainly distort the heck out of them.

The orderly resonances on the left make for a room response that is even, predictable, and easy to work in. Not so with the room on the right.

These diagrams are from The Master Handbook of Acoustics, Fourth Edition, by F. Alton Everest.



Resonances occur naturally, just like the strings on a musical instrument resonate to produce tones.

The ones that have the greatest influence on a room's sound are the primary, or axial, ones, created by only two reflections.

And like the length of the vibrating string determines the pitch, their frequencies are determined by the dimensions of the room creating them - the length, width, and height.



This is called a **waterfall** plot. It looks like a mountain range, with ridges that come down toward us from the ridge line. The horizontal or X axis is frequency, the vertical or Y axis is level, and the Z axis, coming toward us, is time.

The value of this plot is it reveals all the details of a room's ever-present resonances, which are those ridges coming toward us.

In the higher frequencies there are so many resonances, and they're so densely packed together, that there's generally no need to deal with them. Their sheer numbers obliterate any audible effect that they can have on a room's sound.

In the low frequency region, however, there are very few resonances, so it's easy for an isolated one to stand out and color the room's sound. Or for several to be clustered together and do the same. This is why rooms with matching dimensions sound so bad and exhibit so many problems - all the resonances pile up at the same frequency.

These resonances create something like an acoustical graphic equalizer with one resonance like one EQ frequency pushed up. And in the same way, this colors the sound we hear.

And contrary to what some may say, there is **NO WAY** to get rid of resonances. As we saw before, not even splayed or non-parallel walls will do that. But in addition to distorting the resonances as we saw, non-parallel walls also make their frequency shift back and forth, making things <u>really</u> unpredictable, so that it's almost impossible to deal with them effectively.

Ok. Now we have a pretty good idea of how low frequencies work. So armed with this knowledge, what could possibly go wrong in the low end??

Oh, maybe just one or two little...



What, Me Worry??

Major train wrecks!



For starters, the spacing between the resonances is perhaps the most important consideration for obtaining a good acoustical signature in a room. But this is seldom taken into consideration when critical listening spaces are designed.

The best sounding rooms come when we have the most even distribution of resonances in the frequency domain. To get this, we calculate the resonant frequencies and their multiples for all the dimensions, and plot them all together on a frequency graph.



This graph Illustrates what happens when two dimensions are very close together or match.

If we find they plot unevenly, or perhaps clumped together like this, we alter one or more of the dimensions and calculate them again. And again. And again, until we have the smoothest distribution in the frequency domain.

This can be a tedious and time consuming process, but this is one case where the effort is well spent. Once our room has been built, and the dimensions are cast in stone, or in lumber and drywall more likely, we are <u>stuck</u> with these dimensions and what they bring to our room's acoustical signature, right or wrong.

We could also get dimensional ratios from a Bolt chart, or use Fibonacci numbers, the golden mean or other such things, but these are just shortcuts that don't optimize resonances to <u>our particular room</u>. They'll take us a way down the road, but for the best results, we have to put in the work. But the results are well worth the effort.

And if we do this <u>very</u> conscientiously, with a room where we can adjust all the dimensions, we can come close to designing a room that does not need EQ. That's right, no EQ.

It can be done.



We also want to avoid the distorted resonances and their frequency shifting we saw earlier.

To accomplish *all* of this, we make the room a rectangular shape.

Some may call this blasphemy, but it solves these major problems that are uncorrectable once the room is built. By doing this, the resonances become stable, consistent, and predictable, making them far easier to deal with.

The only price we must pay for this, is flutter echo. This actually *is* correctable, and much simpler to deal with than distorted, frequency-shifting resonances. Often with the very same treatments we'll be putting in anyway.



And here's another problem.

Most rooms are built with drywall.

Since drywall is attached to a frame, there are sections of it between the frame members that are unrestrained. Which means the panels can vibrate.

Remember when we were talking about resonant absorbers, and I mentioned panel absorbers?

That's what these become - they're free to vibrate, and that they do, indiscriminately absorbing various frequencies that are unknown because the resonant frequency of an installed drywall panel is difficult to calculate.

But it doesn't end there. Like all things, these panels also have inertia. It takes some time for the acoustical energy to overcome the inertia, and start the panel vibrating, which delays that absorption.

In my testing, the results of these vibrations only start showing up after a delay of around 90 to 120 milliseconds.

And in one particular room, drywall vibrated beyond a half a second.

Now, that's some really loose drywall! Or to paraphrase the old World War Two motto, "Loose drywall sinks studios."

And like peeling an onion, there's yet another problem with this.



Real time analyzers are pretty much the only piece of acoustical test equipment in sound facilities, but they only look at the direct sound, plus *maybe*, some early reflections. They <u>don't even register</u> what's happening later in time!

So this presents a predicament. Because of the inertial delays, the data we really need, to see this issue and to deal with it, **is not available to us** from the one tool at hand - the RTA.

And it doesn't help that we don't even know we need to see this additional information! So whether we're just starting out in audio, or seasoned, Grammy winning engineers, it's not even on our radar.

While our ears have a fine acuity for pitch, timbre, and other things, these delays create problems our ears can only hear, as "something doesn't sound right".

In studios where all we have is a real time analyzer, there's no way to find this problem, even though we can clearly hear its effect.



So now we have at least five out of six room surfaces vibrating and selectively absorbing the lows.

And since they're resonant absorbers, they focus on absorbing specific frequencies.

What do you think this is doing to the low frequency response of the room? Well, I'll show you.


This.

The frequency range is 10 Hz. to 1 kHz., and the vertical axis is 10 dB per division.

The top trace is the direct sound, at 0 ms. The second trace is the very same sound, 240 ms. later - a quarter of a second. There are some profound changes!



A sharp, 40 dB dip has developed at 45 Hz., while the two peaks at 32 and 60 Hz. have hardly decayed at all.



And there's a broad dip developing between 80 and 300 Hz. Do you think this just might sound confusing to the engineer or artist, as it changes this much over a quarter of a second?

Needless to say, this room is not being kind to the sound within it. The vibrating drywall panels are absorbing the low frequencies selectively, and in different ways at different times.

Because we don't understand what happens to the sound during <u>the decay</u>, we don't know where to start looking for the issues which we can so clearly hear. And the engineer trying to work with this doesn't have a tool that will reveal it anyway.

This is the missing element with an RTA, and this missing information is more important than the frequency response that it **does** show.



And moving along in time, we arrive at the response at 480 ms., nearly half a second.

We find more of the same, only worse and worse. We can see the progression of the issues - the two big, enduring resonant peaks in the very low end, with a serious dip between them, that are like party guests who don't know when it's time to call it a night and go home.

And that broad dip of around 15 to 20 dB between 80 and 300 Hz. has developed further. It's pretty obvious that this room won't sound right, but no real time analyzer is capable of revealing these details to us.

I've never heard this topic discussed, and yet it's so important. So let's delve in a little more deeply.

Along with the room resonance distribution and room shape, the next critical element to getting a good sonic performance in a room is...



TIME!



And the problems it can bring to a room's sound.



This is a decay plot from a different room. Similar to a waterfall plot, but with far less clutter.

It illustrates a room's response at various points in time using the familiar frequency response traces, and shows us how things change as the sound decays. These variations from the direct sound may be buried in the decay, but their effects are audible. Pretty obvious here, but well beyond what an RTA could ever reveal.

These <u>discrepancies</u> that the room's acoustical signature makes on the sound between the traces, are what make time one of the most important considerations in diagnosing a room's acoustical problems.

Over time, we can see that some little things can turn out to be a big deal, somethings that look big can turn out to be trivial, some things may not even show up until we're deep into the decay, and some things may bounce up and down strangely, like resonance levels that fluctuate up and down. Which I've seen.



And some things can be terribly consistent, like that peak at 79 Hz.

But there are a few spots where it's still a little hard to pick out the different traces.



So let's look at the response curves one at a time. This makes it easy to see how things change.

Here we have the response at 0 ms., the direct sound. This is a good representation of what a real time analyzer would show us.

The three room resonances are clear to see at 42, 60, and 79 Hz.

Everything else looks pretty normal.



Now we're at 120 ms. Into the decay. The three resonances are hanging in there, and the disrupting effects of the room, are starting to make the response above 100 Hz. uneven.



...including a droop in the region just above 100 Hz., from the drywall panels vibrating.



Now we're at 240 ms., a quarter second into the decay. The 42 and 79 Hz. resonances are still hanging in there, but the 60 Hz. resonance is losing its oomph.

And that droop seems to have changed its mind.



And is something starting to happen around 141 Hz. with that peak?



We've now progressed to 360 ms.

The resonances at 42 and 79 Hz. are diminishing, but still quite present.

And what appeared to be something starting to happen at 141 Hz. has turned out to be a momentary blip - a false start. Perhaps it was a resonance that changed its mind.

And it would appear the droop has returned. This fluctuation could be a bouncing level shift, like I mentioned before.



And finally, <u>finally</u> at the ripe old time of 480 ms., things are starting to settle down. The progression of the traces from graph to graph clearly illustrates how things can change over time, including the droop, which here looks like it's still deepening.



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120 ms.



240 ms...



360 ms...



And finally 480 ms.

This really brings it home, doesn't it?

So if we have the notion that when sound decays, it all decays evenly and at the same rate, it's now easy to see that is **not** the case. Something easy to forget.

If we don't take <u>time</u> into account, we miss a lot of non-linear, uneven changes, the majority of which are focused in the low end, where the bulk of our perception of sound quality occurs.



But there is another low frequency problem that can bite us, one that is related to our speakers' cabinetry.

Up to this point, all this has related to airborne sound, like that which comes <u>out</u> of our speakers' transducers.



Our speakers make the air move, but in doing so, they also create vibrations in their cabinetry.

This physical vibration can enter our room's structure through any direct contact the speaker cabinet has with it. Like from our speaker sitting on the floor, or on a shelf screwed to the wall, or even sitting on our console furniture.

Sound travels through that hard interface very efficiently, and especially so through solid materials - up to 5000 feet per second in steel and some hardwoods. Almost $4\frac{1}{2}$ times faster than in air! Once again, we are reminded that energy is neither created nor destroyed, so this energy cannot be ignored - it will not "just go away."

Which brings us back to the bugaboo of drywall. When this energy from the speaker cabinet travels through the room's structure and reaches an unrestrained section of drywall, this too causes it to vibrate.

This creates a situation similar to the drywall reacting to the energy in the air. But in this case, the unrestrained panel is <u>adding</u> more energy into the air.

So this time we have the wall acting as <u>an additional</u> speaker, instead of taking energy out of the air. And this particular speaker is optimized to be a physical barrier instead of a transducer, so it's going to be pretty low fidelity.

It's not hard to imagine the damaging effect that's going to have on our room's sound, adding this low fidelity sound to the very high fidelity sound coming out of our \$10,000 speakers, not to mention the differences in timing due to differences in propagation speed and overcoming inertia.

It helps to stiffen the drywall and add mass to it. But the best solution is to prevent the energy from entering the room's structure in the first place.

This means providing vibration isolation to the speaker. It turns out this is actually the easiest solution, and provides substantial improvement to our speaker's low frequency response as well. And it's inexpensive, and very quick to do.



This can be accomplished with Sorbothane hemispheres.

They look like rubber bumpers, but they're not.

Sorbothane is a proprietary plastic that came out of the space program, so it's considered to be the best vibration isolation material there is.

We just install four of these under each of our speakers, <u>especially</u> our subs, and the path the structural energy flows through is broken.

This is very simple to do, and provides benefits far beyond the small cost in both money and time spent putting them in. The hardest part actually, is making sure we get the correct ones.

And as a side note, adding Sorbothane isolation pads under air conditioning machinery will also help keep its hum and rumble out of our sonically sensitive spaces.



What, Me Worry??

So...

Now that we know what problems, previously obscured, can and do occur, the next question that comes to mind is...

how do we fix this??



The traditional, go-to approach is to try to fix things with equalization. EQ fixes everything right? Just tune the room and voile! Instant acoustical magic.

But for fixing acoustical problems, this is the wrong tool, being applied to the wrong thing, for the wrong reason. Other than that, it's great!

This is basically applying one error or distortion to another, and hoping they cancel each other out. But in the real world, these things never match up perfectly, do they?

EQ changes what comes out of the speaker, attempting to reduce the stimulation at the problem frequencies.

But as we've seen in the decay graphs, each point in <u>time</u> can have a different response, which would need a different EQ setting. And as we saw with the resonances, each <u>physical position</u> in a room can also have a different response, also requiring a different EQ setting. So with <u>one</u> EQ setting on <u>one</u> speaker, with no capability to track the discrepancies across time and location, the best we can get is a huge compromise.

But there's another problem with this.



With the issue we have here, what we're trying to fix is not a frequency response issue - it's a TIME issue. Our go-to, fix-the-response device doesn't do anything for time. It's the wrong tool for this particular job.

So doesn't it make more sense to fix the actual problem at it's source, the physical room itself, rather than trying to fix it in this oblique, and pretty much ineffective way?

When we fix <u>the room</u>, our design and devices either stop the problems before they start, or fix them in a way that stays relevant and appropriate <u>over time</u>. We can almost say they adjust to the changing requirements of the room.

Or maybe we can turn to our old favorite go-to standby fix - more absorption!



Uh, no, wrong, not this one. Try again...



Ok, this one is a real absorber. But it's still absorption, which means it's still the wrong tool. Throwing absorption at whatever problem comes up is like painting your car every time the engine stalls.

The Secret Life of Low Frequencies	
brucedblack@gmail.com	

Now when we think of dealing with the low frequencies, the first device that comes to mind...



... is the bass trap. As the popular go-to device, it's perhaps the **only** thing we think of. It's automatic! We don't need measurements, we need bass traps!

But as those who know me will tell you...



I don't like bass traps.



And here are the reasons why...



Here's our favorite room graph again.

When you take a measurement of most rooms, this is a pretty good example what you'll see.

As we found, this room has some pronounced problems in the low end.



And specifically they are -

three peaks on the left, at 42, 60, and 79 Hz., an interference notch at 104 Hz., and a group of three lesser peaks at 141, 160, and 185 Hz.

The biggest peak at 79 Hz. is about 15 dB above the average level, while the others range from around 6 to 12 dB above, all significant amounts.

Then we add in the interference notch at around -20 dB below the average level. From the 79 Hz. peak to the 104 Hz. notch, we're talking about a variation of around 35 dB.

Clearly this is not a smooth response at all, in the range of frequencies where we get the bulk of our perception of sonic quality.

So I think we can all agree that we need to smooth this out to get the best, most neutral response to our ears. Bring down those peaks and minimize that interference notch.

So how do we do that? Common wisdom and any number of experts, real or otherwise, would say we need to install some **bass traps**.

But here's my best hack of this graph to illustrate what a bass trap will do to this response. Watch the area around the left four arrows.



A bass trap acts like a bass tone control on an old phonograph, where <u>everything</u> below a certain point gets turned down.

So as I awkwardly try to illustrate here, ignoring where the trace doubles back on itself, we still have the same problems. We've just turned down the entire low frequency range, while keeping the nasty details intact. Throwing out the baby with the bath water.

And if we then feel like the low end sounds weak as a result of this treatment, what will we do? Why, we'll turn up the low end on our room equalizer, which can end up putting us right back where we started. Or worse.

There is also the consideration that <u>true</u> bass traps need to be made from resistive absorptive material, to give them their range.

But remember that this is a material best suited for particle motion sound, while we want to absorb pressure propagated sound.

Meanwhile, the lowest frequency a bass trap will absorb is a function of wavelength; a quarter wavelength, to be exact.

So let's look at the first peak at 42 Hz. With a wavelength of around 27 feet, our bass trap would have to be 6-3/4 feet deep to have an influence on this frequency.

And to be built properly, the entrance of the trap needs to be at the room's surface, <u>in the</u> <u>pressure zone</u>.

How on Earth will we build that behind <u>any</u> studio's walls? Certainly not a home studio, and any business will balk at dedicating this much area to what the bean counters would consider to be turning expensive floor space into unusable dead space. Real ones just take up too much real estate! Especially when you consider that you are <u>still</u> throwing out the baby with the bath water. But there is a better way to control and tailor the low end.

Wouldn't it be much better if instead we <u>reduced the peaks</u>, maybe brought up the interference dip, and <u>left everything else alone</u>? That would flatten out the low end response, removing only the energy that needs to go away, and actually fix the problems. This would provide us with a much more accurate listening experience.

That is what resonant absorbers do.

Using them, tuned to the specific problem frequencies, would give us something like this...



This is my best hack of what an "after" response graph might look like.

It's easy to imagine how much better this would sound, now that those peaks have been individually reduced or flattened, and the dip has been minimized, while maintaining the integrity of the rest of the low frequencies.

This is what we can do with resonant absorbers.

So let's look at that one more time...


We go from this...



...to this.

<u>This</u> is why I always recommend the targeted, tuned low frequency absorption of resonators. You will never see a bass trap do THIS, so you will never see a bass trap in a room I've designed.

Doesn't this just look better??



My favorite is a Helmholtz resonator.

In it's simplest form, it's just a ported speaker box with no speaker in it.

Calculating the port size to tune it is a simple matter, and the design is very flexible.

We can adjust the box's size to fit any available space or unused cubby hole we like, and then adjust the port size to give us the tuning we need. We also adjust the quantity of resonators to determine just how much will be absorbed.

Anyone with minimal carpentry skills and a few tools can build them!

There are other resonant absorbers we can use, like perforated panel absorbers, slat absorbers and more.

But for simplicity, flexibility, minimal acoustical side effects, and ease of construction, you just can't beat the Helmholtz resonator.



So now we can see there are a number critical attributes to the low frequency range, which can be a friend or a foe, a helper or a hinderance.

But with insightful, careful manipulation, we can mold each of them to obtain a neutral, accurate acoustic environment.

Now that we know what they are, and how they work, we can put our new knowledge and tools to work to take control of our low frequency region.



So it is now a brighter day in the world of low frequencies.

We have a greater understanding of all those stealthy little mysteries that plunder the low frequency range, how to identify them, and how to fix them.

We have dispelled some misinformation that can lead us astray, figured out why we can't identify problems we can clearly hear, and shed light on how some things we take for gospel may actually be <u>damaging</u> the room's sound and affecting the quality of the work we do.

Not bad for an hour or so of your time.

You can reach me at the email address shown.



So it may help make things even a bit more clear by scrolling through the graphs quickly.

Remember this is all the same sound, just at different times - no new sound has been added.

This is 0 ms. again...