Sound System Analysis.
The flawed science of measuring the unmeasurable.

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Welcome to the world of the wiggly line.

In order to understand any output of any acoustic, or other, analysis system we need to understand exactly what we are measuring and how it is being measured. This is the most essential step in understanding any result.

The two most common assumptions that are wrongly made are that of the perfect source and the perfect space. If we assume both of those properties then the measurement and result are a simple function of putting out the system and pressing “GO” on the measurement tool. Sadly there is no such thing as the perfect source or the perfect space, the closest we could ever get in is a laboratory, and even then there are imperfections in the perfection. To understand the assumption of the perfect source we need to know the realities of the imperfections of the common source. Loudspeaker systems are at best a poor compromise, the technology behind them is rudimentary and as yet there has been no example of the construction of anything close to an ideal source. In order to generate the required range of frequencies there are contradictions in the design, the loudspeaker needs to be both large and small at the same time. Accurate high frequencies and powerful low frequencies cannot be generated from a single device due to the contradictions in requirements. Engineers the world over spend millions of collective hours pondering this conundrum, but still we end up with the situation where we have to eventually give up and split our loudspeakers up into multiple components, or run components beyond their ideal operating range. This leads to two fundamental flaws that exist in all loudspeakers. Internal interference and external interference.

Internal interference is where the physical properties of the individual components of the device cause problems in the propagation of the generated sound, this is usually the point where drivers cease to work as a piston and enter various forms of break-up or where we begin to get geometrical anomalies where at certain listening angles we get propagation from one part of the loudspeaker cone (or diaphragm) arriving out of time from that from the other side to the extent that it will interfere with itself and produce a complex interference that then colours the response. In horn loudspeakers we can also get problems caused by interference in the propagation of the wave fronts along the horn by either geometric changes in the horn itself or obstructions within the horn. [1] [2]

External interference is where we have multiple drive units operating as though they were one. In this case it is almost impossible to co-locate two drive units, there is almost always some form of dislocation of sources and in the case of almost all frequency dividing systems needed to make multiple drivers work as one there will be a part of the frequency band where we have simultaneous generation of the same frequency from two very different devices. Where the intended listener is not a perfect precise single point in space we will then have the problem of interference between the sources at different geometrical positions from the multiple drivers where there is only one point in space that is equidistant from both drive units all other points are of varying distance. As sound significantly travels in time as it travels in distance we will obviously have time-shifts between sources at any point other than the equidistant point, thus the two devices will inevitably begin to destructively interfere with each other at points in space entirely dependent on listener position and generated frequency. *Sound is very slow, recently a person, without vehicle or means of propulsion, managed to travel faster than sound and survive (Felix Baumgartner – 822mph free fall)*. As sound is so slow and audible wavelengths reasonably short we can see that at 1,000Hz a single cycle is 344mm, so we only need for there to be half of that distance difference in path from the two loudspeaker drive units to our ear (or measurement microphone) for there to be absolutely 100% cancelation between the sound waves. (Half wavelength = 172mm where one drive unit's positive pressure matches the other drive unit's negative pressure resulting in zero pressure) This example is only at one single frequency. The interference is greatest at crossover frequency, but with shallow slopes, or sources of more than two frequency bands we have many more instances where we have this off-axis destructive interference. This interference is greatest in the axis along which the drivers are arranged.

So, as we see, with a real world source with both the single drive unit polar response anomalies and inter driver interference we will have almost no two listening positions where we could possibly achieve the exact same acoustic analysis measurement. There will be variations between minor differences and huge anomalies depending on which position, or groups of positions we choose to measure from.

At this point we are still assuming that we are measuring in a perfect space. Our ship is already sinking and we haven't even gone to sea.

Talking about sea, that is a close analogy to the next issue.

The room. In all real world applications we have three properties that most affect our measurement. Firstly the source, then the space it plays into and finally the position we listen from. We've already dealt with the variability of the source, now let us look at the other two.

The room itself becomes more problematic the further away we go from an anechoic space.

In a truly anechoic space, or outdoors, we principally have to deal with only the source anomalies and the weather. As soon as we enclose our source with any degree of reflective surfaces we send our sound bouncing around until it is either absorbed by the air or by the surfaces it is hitting and transmitting into to the point at which it becomes inaudible. As it travels along its merry way, being reflected, it begins to meet itself again (or at least the next cycle) and will interact with that sound as it passes causing either cancelations or summations at those points. Corners, holes such as doors and windows, or incomplete boundaries begin to "slice up" the wave-fronts and send them in different directions creating ever more complex patterns of interference as the journey continues. As the points of interaction are dependent on the propagated wave energy we will see a different pattern of summation and cancelation for every frequency generated as each frequency generates a different wave length. Without taking more complex issues into account we can already see that even in a relatively small room we are getting to the virtual realms of a space with infinite variation. No two points in any enclosed space that is not totally anechoic will have the same exact frequency response when considering the reflected energy interaction. Careful response analysis generally backs up this concept.

So far we have looked at two independent issues, in each case we assume the other to be a perfect theoretical example. In our room, we have assumed that we had a perfect spherical propagation before it started hitting the walls, and with our loudspeaker we simply assumed it was propagating into free space. If we take on the real-life situation of a loudspeaker in a room the complexity and potential variation of both the source and reflected energy becomes incalculable. It becomes very obvious that every single point in space will have its own personal acoustic signature which is dependent upon multiple factors. We haven't even taken into account the global modification factors such as air temperature, pressure, currents and humidity. Changes in air density and water content can modify the speed of sound, which in turn changes the wavelength of fixed frequencies. This in turn will change the physical points in space at which interactions occur. We now suddenly have variations on top of variations of a variable source. It is clear at this point that it is really impossible to have a simple single global value of anything that represents any form of representative listener experience of the situation before us.

The best answer at this point is quite simply 42. [3]

It may as well be.

This all sounds rather pessimistic, surely we have been able to measure the loudspeaker in a room, there are countless times when we all have seen and used such measurements and they seem to have worked.

Well, there is nothing at all wrong with the measurement systems, virtually all of them, even some of the cheapest will give very accurate results far beyond the realms of very expensive analogue systems many years previously.

So if the measurements are all good why do we have a problem?

Just as in the answer being 42 [3] which it may as well be, we better need to understand the question before we can understand our answer.

When we place a measurement microphone in a "listening position" we ask just one very simple question. That question is "What do we hear at this tiny point in space?" now depending upon the function we perform upon the signal from the microphone we can get a variety of results.

With a standard common RTA measurement we get the absolute total frequency-amplitude response of all energy entering the microphone at that point in space. There is nothing else to that measurement. Whatever is going into the microphone is placed on the display irrespective of what it is where it came from, or when it was generated. This very crude method of measurement is quite blind to many aspects of the source, or sources. We simply have a set of *(usually 31)* band limited SPL values.

Happily, as listeners we each own a very powerful audio processing device known as "The Brain" which coupled with two very precise atmospheric pressure sensors known as "The Ears" is capable of processing a vast amount of data from the modulation of the atmospheric pressure at the ear at any given point in time relative to the pressure at a previous point in time. Our immense sensitivity to temporal variations coupled with massive processing power allows us to easily separate direct sound from reverberant sound and sound from position A from sound from position B There is an immense amount of work out there on such subjects, but the general consensus nowadays is that the response measured on an RTA type device is not representative of the perceived response as we possess the ability to differentiate between the direct and indirect sounds so long as they are adequately distinct from each other.

The advance of modern processing power, coupled with rapid development of the application of mathematical functions upon captured acoustic data has now enabled the easy and cheap production of more complex measurement devices. These devices are more capable of extrapolating direct sound from reflected sound when supplied with a known source test signal. This is known as a windowed FFT (Fast Fourier Transform) measurement. So long as the system operator manages to input an adequately representative window size to the measurement system it is, once time aligned itself, able to extract the source signal from the mass of reflected energy in such a way as to remove the room from the measurement. This is by no means an exact process as different frequencies require different length time windows which should be correctly set to allow only the required data through and at lower frequencies there is still debate as to whether such data can be correctly extrapolated, or is even extrapolated by our brains when listening. Even if the room energy is extracted in a way that is representative of our own auditory systems we are still left with a complex and varying sound field that is the directive anomaly of the source loudspeaker in such a way that we have a massive and complex variation in our sound field over the listening area. While a windowed FFT measurement will to some degree represent the direct sound it still will not give us any clue as to what everybody is hearing.

We are however getting much closer to understanding our own version of 42 [3].

It is becoming increasingly obvious that the simple wiggly line we see on our graph can never be "**THE** response" it is simply "**A** response" both our source and our space have an almost infinite number of responses all dependent upon multiple factors. We are looking upon something that is almost as complex as the global climate and trying to express it in the form of a single number of line on a graph. We still yearn for the simple line that will give us the representative response of the system in the room, but the interaction of all the components is so immensely complex and results in a hugely complex sound-field that there simply isn't **A** response to represent the system. Work carried out for various published papers clearly shows that applying selective amplitude correction to a source loudspeaker to tailor a response at one point in the room can have inversely productive results at other points in the space. [4]

There are equal flaws in looking to use a plot that is the average of multiple tiny pint sources in this massively complex space. No one single listener could ever experience the sum of the average response – not unless their ears were as big as the averaging area. It has been proven that as with the individual point source measurement, a collection of individual point source measurements is subject to similar levels of variation dependent upon the position of the component measurement positions [5] Additionally it has been shown that the average response cannot be shown to be representative of any of the individual listening points so cannot be a valid starting point for application of system processing [5].

Well where does all this leave us?

For sure, many of us who have worked for years on system tuning already know that the wiggly line is just a guide, a sort of confirmation tool that what we hear is actually there, somewhere, and more importantly tells us an easily quantifiable number for our intuitive judgement. It certainly isn't a substitute for the ears and brain of a good system tech, or walking the room with our ears "open" Analysis systems can be great tools for time domain analysis, system driver time alignment and basic guidance on where an issue lies. They certainly aren't even remotely capable of giving us an accurate representation of how a system performs in a space, or even as a whole in isolation. They are only ever capable of telling us what is happening at the microphone diaphragm at that point in time.

The world's most accurate sound level meter can't even tell you how loud a fixed tone is in a room, tests have shown up to 15db uncertainty from measurements within the same cubic metre from complex sources at a fixed tone. The industry is still examining itself and trying to figure out the way to get a single number from a complex data set.

Any result of analysis of an acoustic space that is represented as a simple number or graphical line must be taken with a huge amount of caution, claims that "This is the result" without further qualification of that result are at best for entertainment purposes only, they serve no function and would most certainly not withstand solid scientific scrutiny.

[1] http://www.klippel.de/uploads/media/Measurement_and_Visualization_Cone_vibration_06.pdf