ACOUSTIC POWER LAB

APL

Equalizing loudspeakers

Acoustic Power Lab
http://aplaudio.com

This is a well-known problem for equalizing any system with the linear (usually complex) transfer function K(jω) by use of a correcting system (circuitry) with the transfer function 1/ K(jω) that equates as K(jω)*1/ K(jω)=1 unity transfer.

For some this is all very "ordinary" and long-ago resolved and there are no problems making the electrical and acoustic waveforms match in shape as closely as needed.

But it is not such a simple case with loudspeakers. The above holds true for 4-pole systems (2-poles input, 2-poles output). But a real loudspeaker is never a 4-pole system with lumped parameters. It is a distributed parameter system (where the size of system elements is equivalent or larger than the operational signal wavelength in those elements) and one cannot define its 2 output poles. To simplify that the sound pressure "on axes" is representing the output is fine for very few specific cases when the loudspeaker membrane is working as a true piston and there is just one membrane (ideal point source). The loudspeaker industry is close to achieving piston motion but not quite there yet.

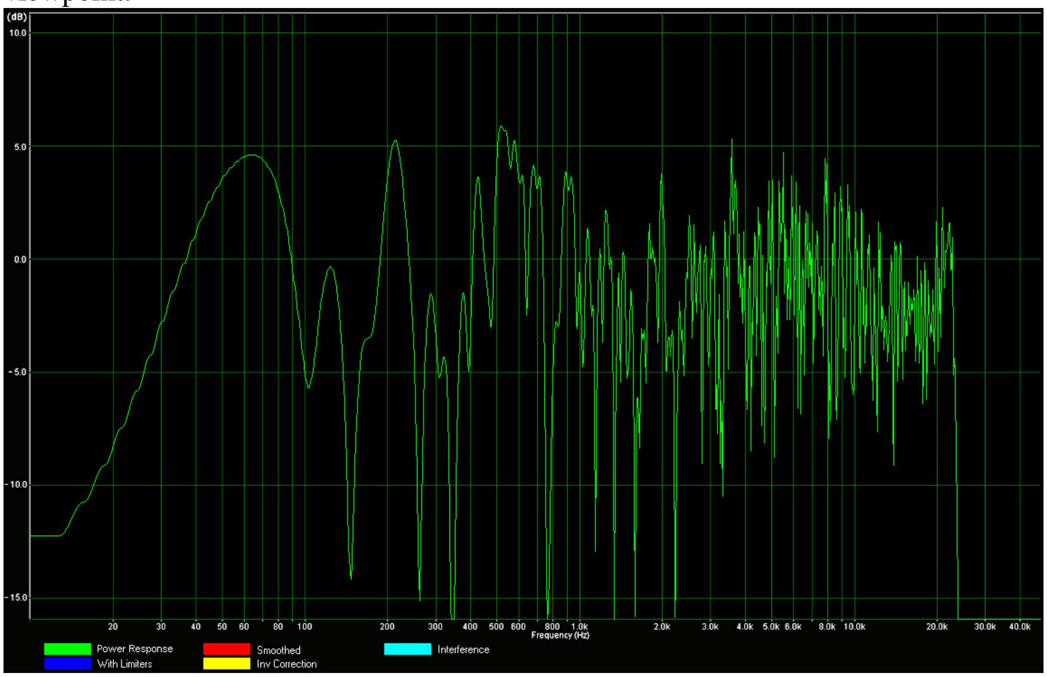
And the fact that it cannot so simply be proved by the fact that there is no working solution that takes an "on axis" response, inverts it, creates a filter with such an inverted response and that is all. Anyone who has tried to do this knows – the result is unusable. Why? Because the loudspeaker is not a 4 pole system with lumped parameters.
While this might cause one to get discouraged and just give up, things are not as bad as they seem.
The described situation holds true if we are trying to create some calibrated sound field may be for some scientific purpose by use of a loudspeaker. If we use the loudspeaker to create a sound field for „human perception" it doesn't look at all impossible because of the valuable properties of human sound perception - „party effect", time selectivity and the ability to abstract from wave interference (we can listen to a two-speaker (interfering) stereo system and not be disturbed).
I propose that it is possible to build a loudspeaker that, while not ideal for scientific purposes, can nevertheless create an uncolored „sound image" of any live sound source that "looks like" that live sound source exactly.

To solve the audio industrys basic task (goal) (especially for sound reinforcement applications) – to achieve a situation where the listener does not perceive the presence of a sound reinforcement system and is „thinking" that the exceptional performer's performance and the exceptional acoustics of the hall are „responsible" for the great performance. [Ahnert, W. and Reichardt, W., Grundlagen der Beschallungstechnik (Foundations of sound reinforcement engineering) (Berlin: Verlag Technik, 1981)]

Therefore we need a solution that works with real loudspeakers, to evaluate and improve their performance and to apply equalization as accurately as possible by exploiting the above-mentioned human sound perception properties. This means that our corrections must exactly represent the perceived problems and pre-distortions introduced by the correction circuitry (equalizer) are compensated (neutralized) by the distortions of our loudspeaker. If this is not so, we will introduce new distortions and not have solved our problem.

Therefore the idea of room correction by applying some equalization before the loudspeaker will just be creating distortions of loudspeaker sound and does nothing to the actual properties of the room.

Sound pressure picked up by a measurement microphone in steady case (we wait a long enough time for all transition processes to be completed)– was and is (for some) the final authority because it looks like a very „accurate" (detailed) from a scientific viewpoint.



But to make real equalization decisions we must decide to somehow process it, to smooth it out (at least in how the mind perceives the sound). And we lose important informative details about the loudspeaker's performance this way.

On the other hand, our hearing perception is so advanced as to be able to detect and extract information about the main sound source away from disturbing ambiance (party effect) so that, by use of its timing selectivity, it detects and focuses on the „main" signal in time - which was very important when we were living „in the forest" and facing bears and we haven't lost this evolutionary ability yet.

Example of an ordinary stereo system.
If we sum some signal with its delay electrically, we get (and hear) a comb filter, or flanger if the delay is variable. If we feed the signal and its delay into different loudspeakers – no comb filter for our perception (just panning between speakers if the delay is changing) but the comb filter is for microphone „perception".
Comb filter usually is called wave interference or just interference. Let's use this term.

Example with recording acoustic balance.
It is a well known effect in the recording industry that the acoustic relationship (direct sound/reflected sound) is much worse for microphone „perception" than for human perception (you must put a microphone much closer to the performer to obtain the same balance as listening to it „live") .
We may explain this as unordinary for technical means timing and direction selectivity of human perception that is not repeated by any sound pickup, recording or processing means.

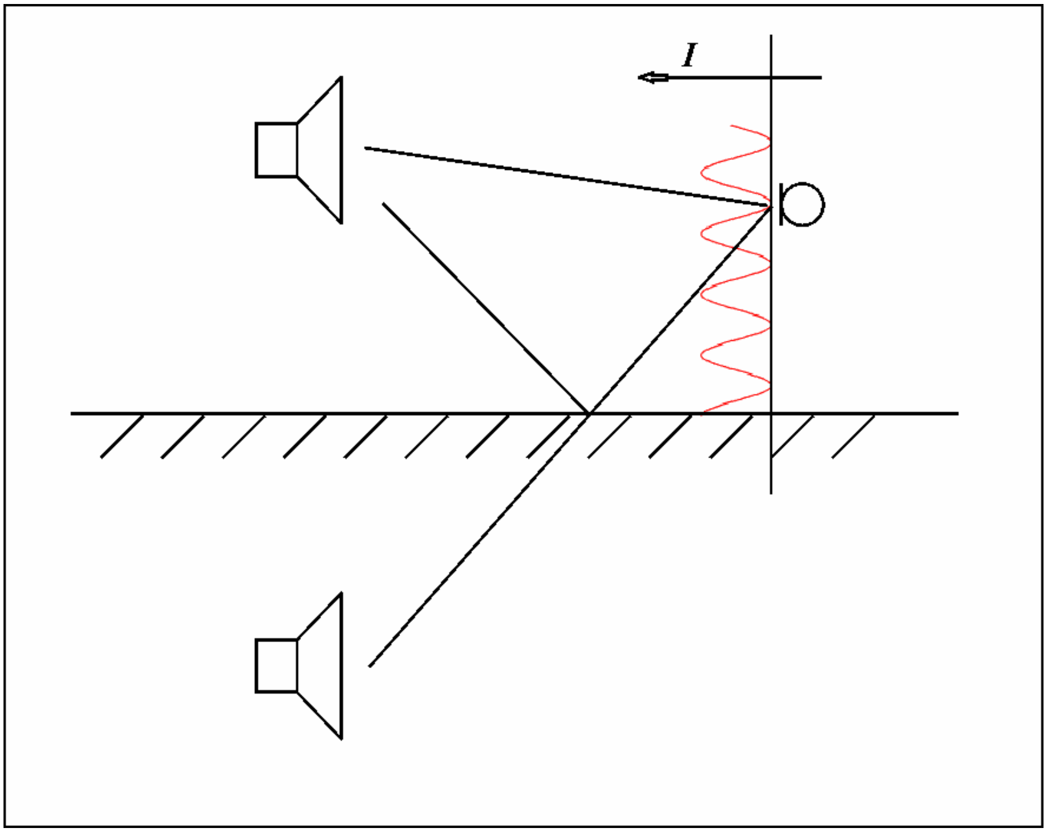
All trials doing equalization based on measurements that contain interference (are not interference-free) have been unsuccessful because the pre-distortions and distortions are not compensating each other for our human perception and even for different measurements of microphone position as the wave interference picture is changing dramatically and that does not depend on whether such interference is created by not coherent (not piston motion) movement of the loudspeaker membrane, by sum of two (or more) loudspeaker fields working at the same frequency band or by some reflection.

The time selectivity of our perception that separates direct sound from reflection (delays) (described with „party effect", „stereo system" and acoustic balance) tends to suggest that we, as result of things mentioned, do not perceive interference as a disturbance. And it leads to the idea of using interference-free measurements for loudspeaker evaluation.

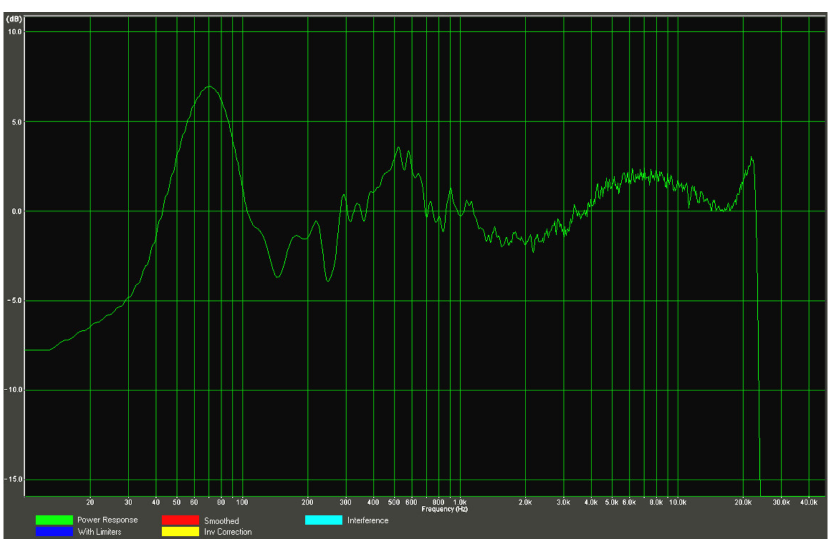
I present two solutions that are free of interference.

First – let's work in the SOUND POWER domain
Second – let's use exceptional time resolution to emulate our perception

Power domain.
As mentioned, you can not find loudspeaker „output" poles. This case is very similar to microwave engineering that uses wave guides. Usually no one talks about voltage and current in wave guide or any other microwave units, especially antenna, but use Power. We find many analogies in loudspeaker and antenna engineering.
You can also find the use of *power* in scientific articles about fundamental principles of loudspeaker efficiency. The value of the emitted sound power is used to describe some sound sources that usually are not loudspeakers (a chain saw for example). And such evaluations are done in Reverberation chambers (as opposed to an Anechoic chamber)
But it is hard to find a tool that could evaluate loudspeaker performance in the *power* domain for our everyday usage in field applications.
The beauty of the power domain is that it does not try to describe the sound field at some point in space. It describes the sound source itself.
Let's illustrate.



The loudspeaker is placed at some height from the floor and its emitted sound field is creating the interference picture caused by two sound sources – the real loudspeaker and its mirrored image. The interference picture for some particular wavelength (frequency) is shown as a red curve. If we accidentally put a microphone in position of null we may decide that the loudspeaker is not emitting any sound at this frequency. But if we collect information about the sound intensity from many points and then integrate them we will obtain information about the sound power emitted by the loudspeaker. We can do this for multiple frequencies to obtain the Sound Power Frequency Response of the loudspeaker on the test.



The practical value of this curve we should test as usually we do with any equalizing decisions. Let's return to history. Many decades ago we had two „knobs" – Highs and Lows. Tweak these knobs, listen to the effect and decide to use this new particular „setting" or to return to previous one. Then we got 10 knobs (1 octave EQ) and did the same process. Then we got 31 knob (one third octave EQ) and did same trial process.
Then we got spectrum analyzer with 31 LED strips – we looked at an analyzer and tried to move some of the 31 knobs, listen to the result, return back if the result was not satisfactory.
We can do a similar thing with available SPFR– let's create an equalizer that exactly „follows" SPFR and listen to the results ...

That was done 10 years ago with a very important observation – the result is good for any case - you never discard such a result

As a result the practically usable (especially for field applications) solution has been proposed to the industry for almost 10 years now by the author of this article and is already being applied by a number of well-known names in the industry - Community Professional Loudspeakers, Panasonic, Kenwood, JVC, NEC, Toshiba, Hitachi …

It is exploiting the sound power pickup (registration) principle that incorporates integration of sound intensity (that is sound power traveling thru some surface) around the sound source and is a well-known method for measuring the emitted sound power of industrial devices.

After equalization based on SPFR field measurements was proposed in May of 2005 some industry players „reinvented" it and began to propose a semi-solution of „averaging" of multiple measurements with late expression that must be done in the POWER domain. John Murray described this in his article "Exploring Converging Techniques For Tuning Line Arrays"

http://www.prosoundweb.com/article/exploring_converving_techniques_for_tuning_line_arrays/av/P2/

But as this solution was and is for the sole purpose of „looking" at it and maybe to turn some knob, it not a substitution for SPFR measurement using 100 ... 200 measurement points (taking 30...60 seconds of time) and sequential creation of an uncompromised equalizer filter.

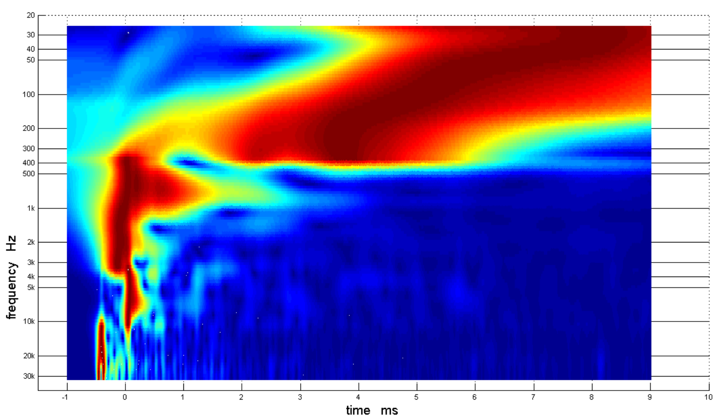
But practical use of SPFR measurement for equalization has two drawbacks – 1) loss of frequency resolution caused by time windowing for very low frequencies (20...30 Hz) where complicated subwoofers with significant delay are being used, 2) need to take into account loudspeaker directivity for the usual HF bands found in real loudspeaker systems. There are ready-to-use solutions to deal with these effects but they require additional skills from the operator and time to implement them.

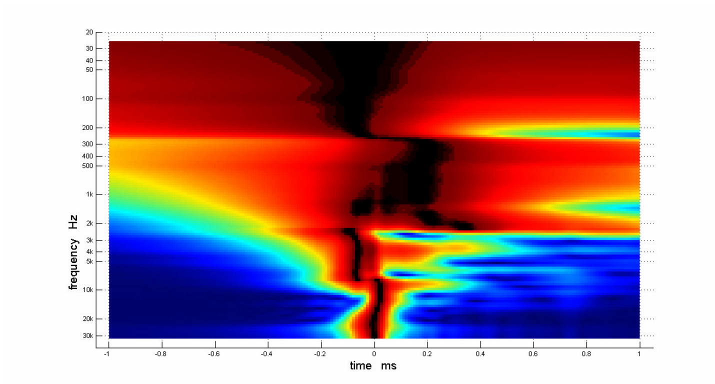
Use of high time resolution.

Let's call it Time Domain Analysis (TDA) for the nature of such work in the time domain as opposed to the frequency domain as for most of the tools already used.

Almost all audio analyzers use Discrete Fourier Transformation (often called Fast Fourier Transformation for data size that is power of 2) and derive (calculate) all their information (curves) from data obtained through FFT.
But you must supply FFT with a sequence of samples –a block of data. This block must be quite large to have a usable frequency resolution. But with large block size we get very inaccurate timing information that is comparable to that block size. The Time Delayed Spectrometry (TDS) is its implementation with „waterfall" graphs as a result.

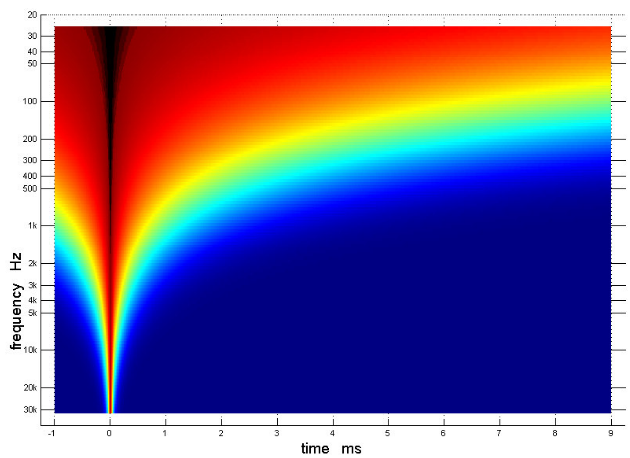
This brings us back to history to when the first audio spectrum analyzers were built, incorporating a number of Band Pass filters, detector circuitry and LED strips as indicators.
And also, it brings us back to our human hearing that is using many resonators, BPF, (mechanical?) to analyze sounds that we perceive.
The processing of Band Pass filters outputs gives us very interesting, high resolution timing information that allows to see how the signal energy of different frequencies travels thru a system (or a loudspeaker, particularly) and to see (as a result) frequency-dependent delays directly in graph form and in very high resolution. And please don't worry – the Heisenberg-Gabor limit is not broken. We are just a bit closer to this limit.
From the Heisenberg-Gabor limit we know that dF*dt=CONST. For FFT analysis this constant tends to be 1 (or some part of 1). But for TDA, it is about 1/20 … 1/50 - about 10 times (or more) better than FFT.



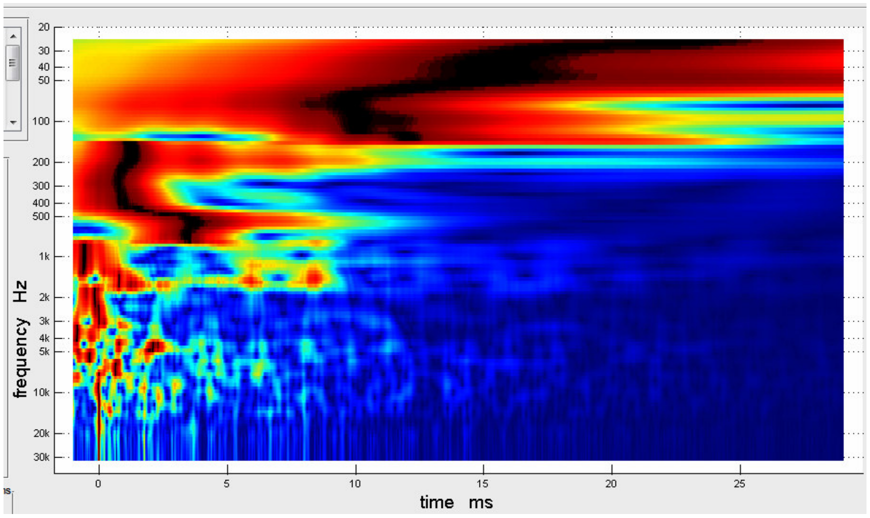


The two images above show examples from real life.
First – some 4-way loudspeaker system with band's delays slightly out of tune. .....
Second – a pretty well-tuned system and high zoom in (+- 1 ms) is showing the details

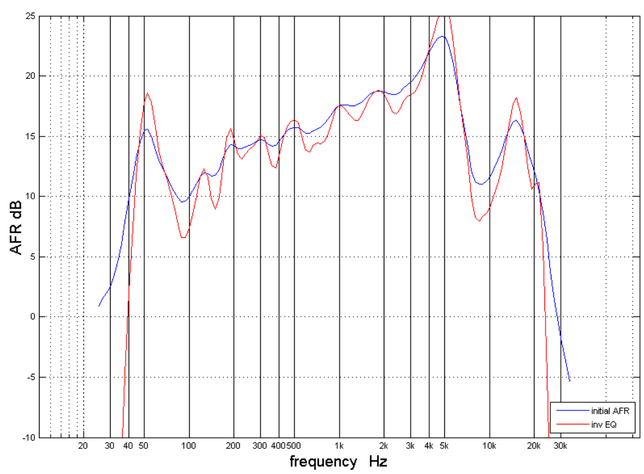
In an ideal case the picture should look as follows below:



The time resolution is so high that we can see the effects of not having „piston motion" in a loudspeaker membrane. In that case different parts of the membrane are moving and emitting sound in different phases creating at some particular point in front of the speaker and at particular frequencies the „out of phase" effect. But there is no such out-of-phase effect at this frequency in other directions. If a loudspeaker is surrounded by some reflective surfaces, the microphone picks up the reflection signal (for that particular frequency) from the direction that is not „out of phase" and shows it as delayed. This is true especially for an interior automotive environment.
The true electrical crossover „out of phase" is displayed in the same way as a strange „reflection" on the crossover frequency.



The maximum of the detector's output signal represents not only time of arrival but also the magnitude (power) of the arriving frequency components. And as this magnitude represents one particular moment in time, it is in a way free of interference (not absolutely) from the delayed signal (reflection) that can be seen later on the graph time axes.
This allows as to put the time selective Amplitude Frequency Response of our system to the test.

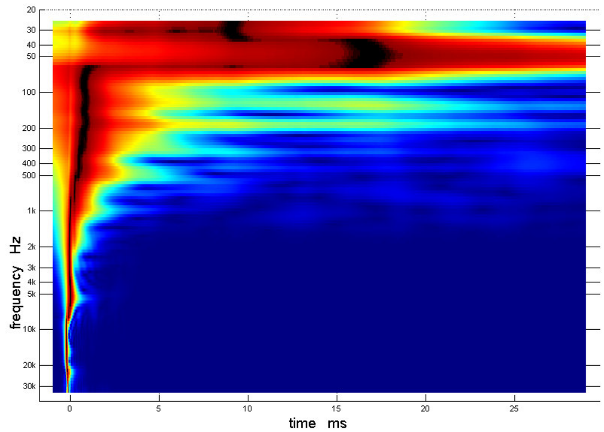


And it tends to use this to make an equalizer.

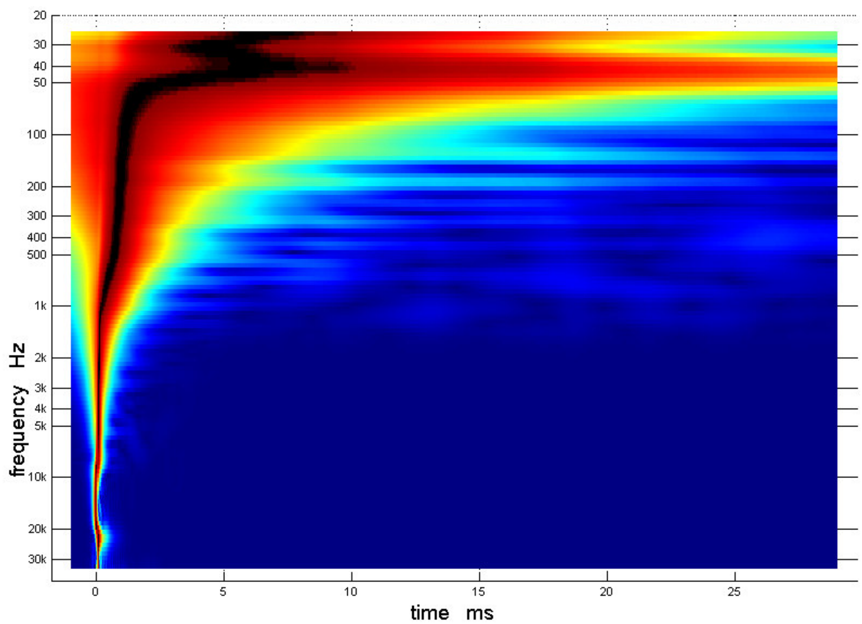
The first tests of such an equalizing approach have shown very promising results as seen on measurement graphs as well as heard in listening tests.

The TDA graph is showing serious improvement in phase/delay caused by use of a minimum phase equalizer that not only removes boosts on AFR but also corrects phase response.

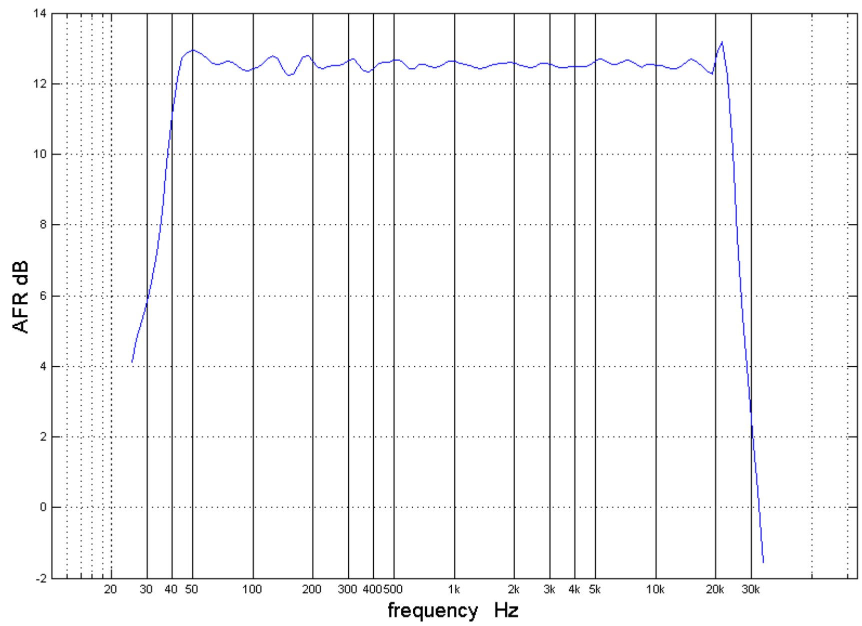
One sees phase/delay problems as non-symmetrical slopes (to positive and negative time) of TDA detector output curves for a particular frequency as shown by the yellow color levels.



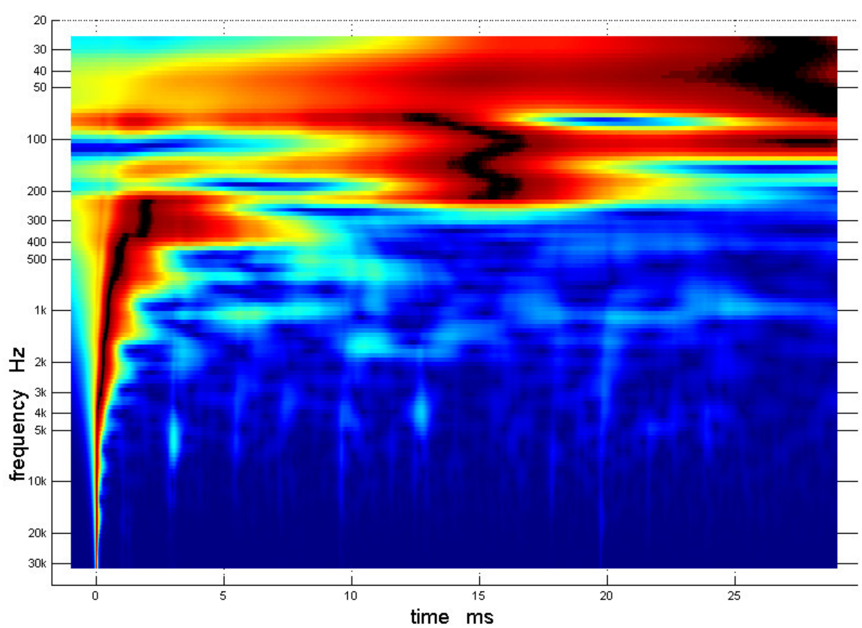
There is considerable improvement in the TDA graph after equalization



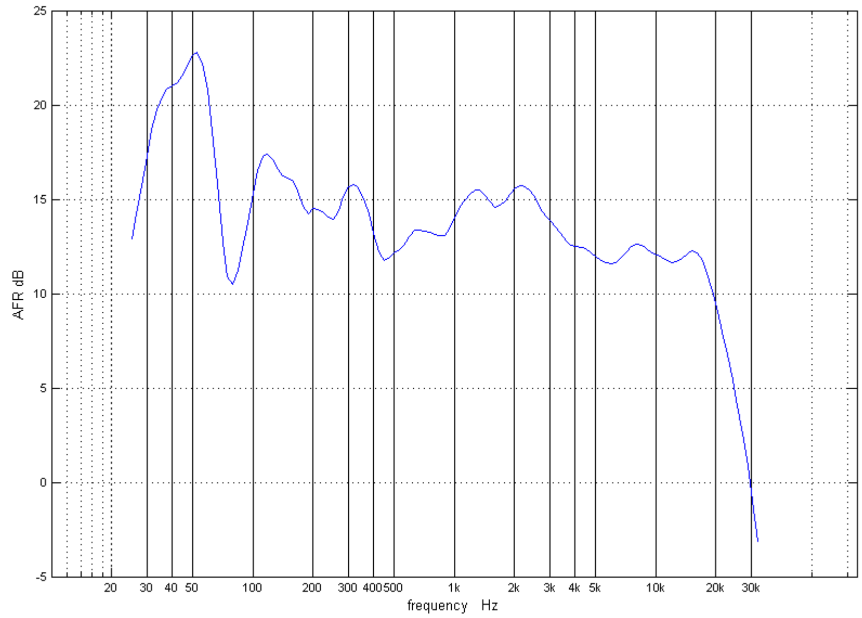
And equalized AFR



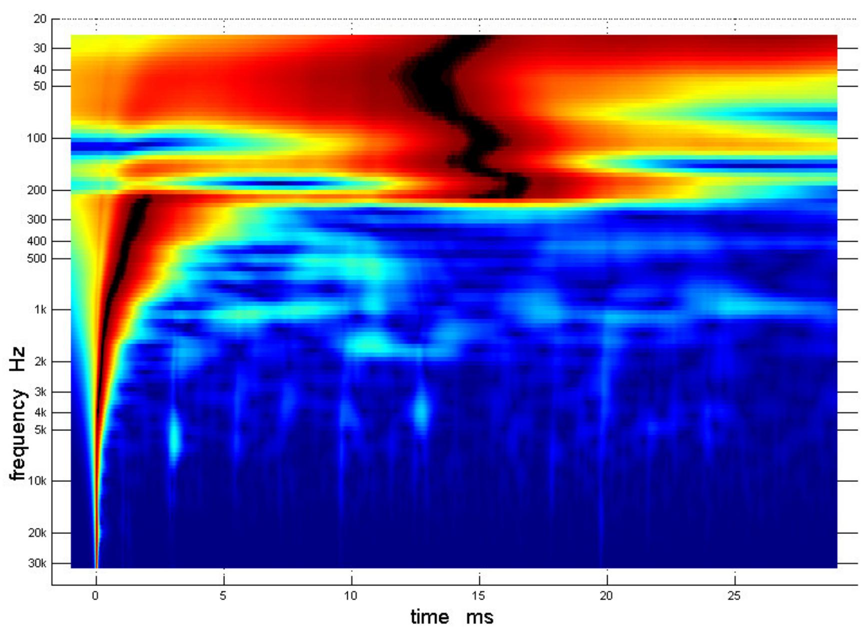
If a loudspeaker system is 3- or 4-way it is a challenge to keep the crossover circuitry well tuned. It is easy to make serious mistakes if you are doing this "blind".
This happened with a large (and expensive) 3-way studio monitor where incorrect design decisions introduced a 20 ms delay in an LF band that should not have been in a loudspeaker intended to work as a studio monitor.
Let's look at that example.
The TDA graph before any equalization.



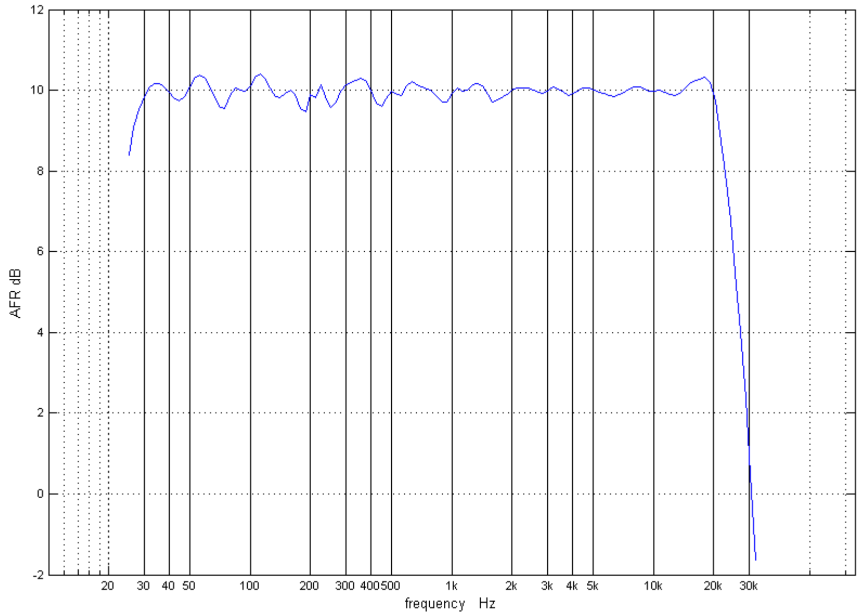
AFR



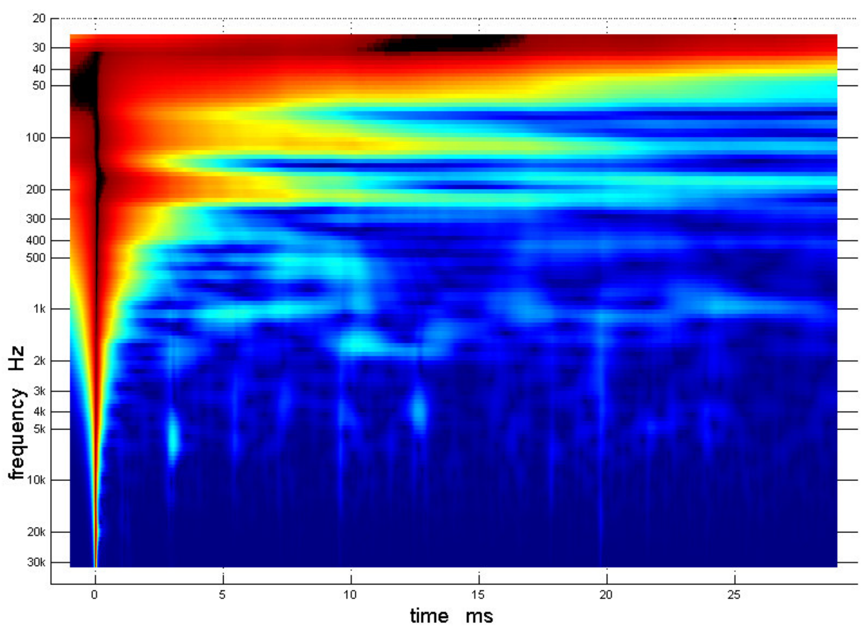
The TDA graph after minimum phase AFR equalization was applied



AFR

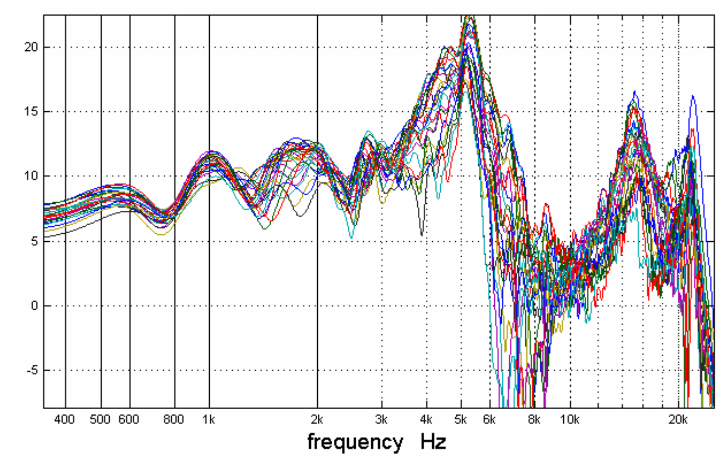


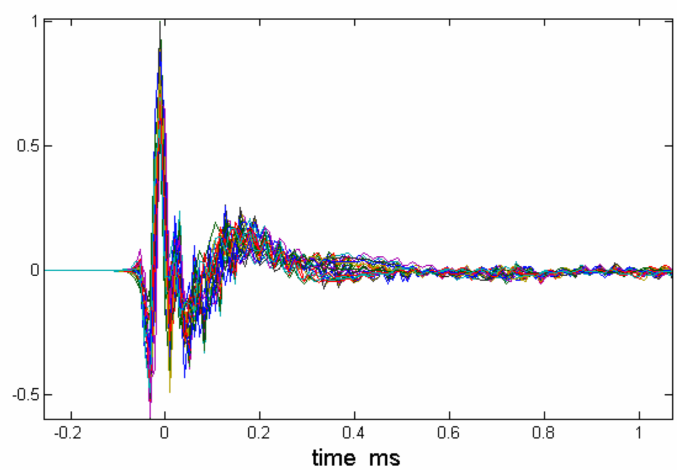
Now is the time to apply phase/delay correction based on DFR.
The TDA graph after both equalizations



The equalization based on TDA analysis compared to the one based on SPFR shows much better resolution and accuracy in the very low frequency range (caused by the time window used in SPFR measurement) and a much better result for near “on axis” for systems with problems in directivity (narrow beam in HF band)

The information for TDA analysis may be taken just from the measurement point for systems (usually studio monitors) with well equalized directivity (membrane piston motion achieved for full frequency range)
For other systems, a certain number (25 for example) of measurement points should be used that are at some degree from the main radiation axes of the loudspeaker. Some partial SPFR evaluation should take place in the way needed to deal with directivity problems (non piston motion of membrane) of the loudspeaker. But it doesn’t “cost” as much to use a multipoint measurement for any system. The first version of the TDA EQ software works exactly in this way.



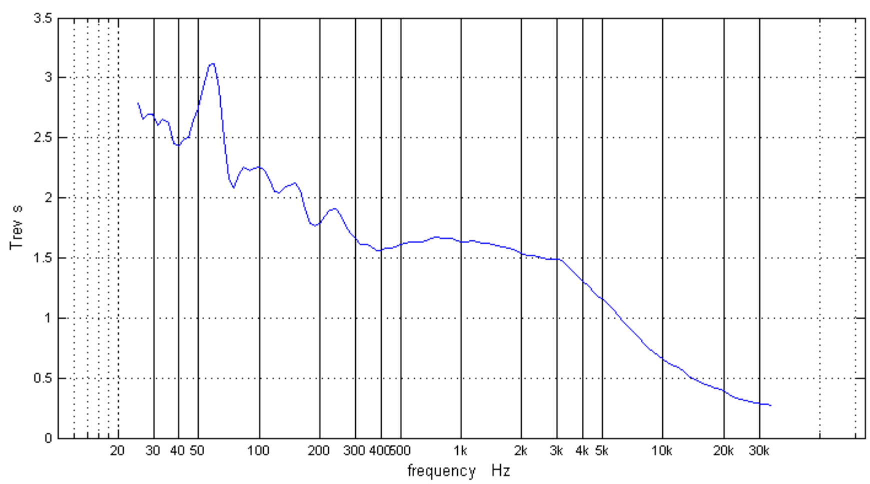


EQ for room/hall
What we can do?

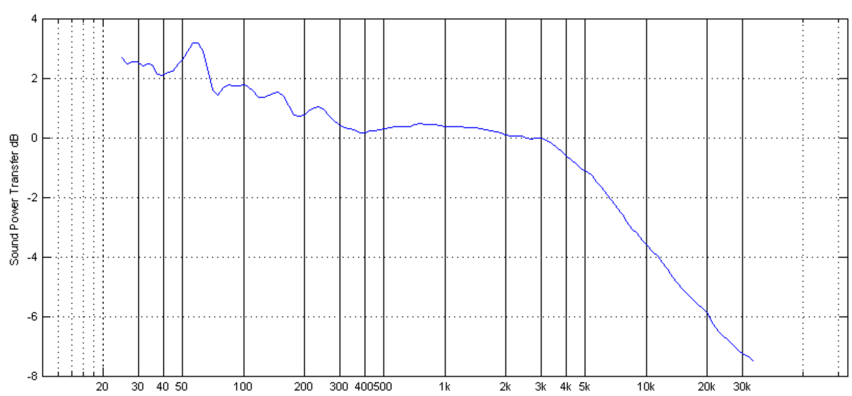
There is an understanding in the industry that wave interference can not be equalized by introducing pre-distortions in the signal path before the loudspeaker and must be left as-is for room/hall evaluations. But what about other hall parameters that don't depend on a current point in space but describe the hall as a whole.

Sound power absorption/accumulation properties of the hall.

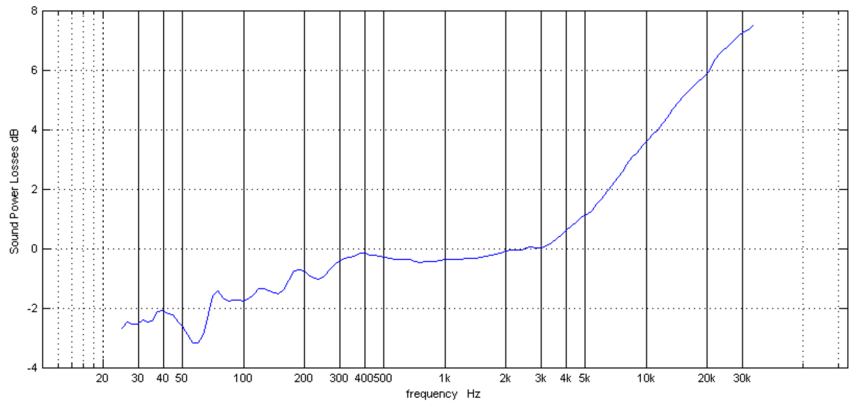
Reverberation time has the direct connection to the coefficient of sound absorption (losses) in any given hall. And that absorption coefficient *a* (relationship Pa/Pr – absorbed sound power versus sound power accumulated in the hall) of the room is inverse proportional to Reverberation Time. $RT \sim 1/a$
And Sound Power Density (SPD) created by some sound power source with power P in the hall has relationship SPD ~ P/a or SPD ~ P*RT or (SPD/P) ~ RT
We can call SPD/P a sound power transfer coefficient and, as usual with transfer coefficients, we can describe it on a logarithmic scale as 10*log(RT/RT0) where RT0 is some freely chosen reference RT.
Let's look at an example.
The reverberation time frequency response of Benedum center



Let's calculate it in logarithmic scale to see it as a sound power transfer coefficient



and invert it to see it in terms of sound power "losses"



This is how the hall equalizer FR should be to create a flat Sound Power Density Frequency Response in the Benedum center from the sound source (loudspeakers) with a flat Sound Power Frequency Response.
I trust any readers experienced in real hall conditions would agree that he is using more or less similar eq for work in the hall.
But now it is possible to create EQ that exactly reflects the properties of the hall. Why wouldn't you do this?

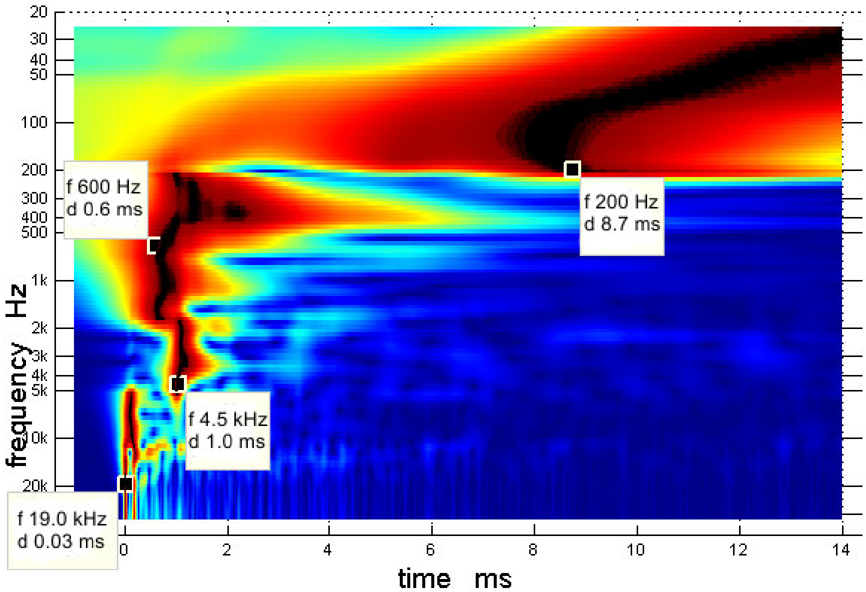
In conclusion.
With this article I hope to have made some contribution to the field of Sound Power usage, as mentioned by John Murray in his article "What's The Measurement? Understanding And Properly Using RTA & FFT"
http://www.prosoundweb.com/article/whats_the_measurement_understanding_and_properly_using_rta_fft/P1/

And the use of interference-free measurements described above will be just as free from the "Big three" errors Murray lists there.

I also wish to point out that this direct synthesis of equalizer filter as described is what gives a new level of quality by freeing one from extended parametric equalizer "tweaking" to get a particular curve and losing focus on the main task. If you need 5 parametrics to make a correction for some one (on a frequency scale) problem, you are losing too much energy trying to find those settings and so losing focus on your result.
Let's also use CAD (computer aided design) for EQ.

A loudspeaker system must be tuned as accurately as possible before any use of overall equalization.

A high time resolution of TDA allows certain system tuning jobs to be done much more effectively and error-free.
Joan La Roda in his article "Phase Alignment Between Subwoofers And Mid-High Cabinets"
http://www.prosoundweb.com/article//phase_alignment_between_subwoofers_and_mid-high_cabinets/
It takes 5 pages to describe all of the techniques required to do that.
Now, just one sweep and the TDA graph shows you everything you need to do.


f 600 Hz
d 0.6 ms
f 200 Hz
d 8.7 ms
f 4.5 kHz
d 1.0 ms
f 19.0 kHz
d 0.03 ms
frequency Hz
time ms
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