Understanding the “state of the art” of Digital Room Correction

with

Mitch Barnett

Overview

- What is Digital Room Correction (DRC) and why do we need it?
- DRC is a specialization of Digital Signal Processing (DSP). DSP is everywhere.
- Brief history lesson: How DSP revolutionized the Pro audio industry.
- How do we model loudspeakers in rooms using DSP.
  - What is important to the model and why.
- How do we design the ideal transfer function for any given loudspeaker in any given room.
- Examples plus hands-on walkthrough using DSP Filter Designer's
- Conclusions

What is Digital Room Correction (DRC)?

- Digital room correction (or DRC) is a process in the field of acoustics where digital filters designed to improve the unfavorable effects of a room's acoustics are applied to the input of a sound reproduction system.

- DRC is usually used to refer to the construction of filters which attempt to **invert the impulse response of the room and playback system**, at least in part.

- Modern room correction systems produce substantial improvements in the time domain and frequency domain response of the sound reproduction system. Wikipedia entry.

- The goal is to restore the sound quality so there is no frequency or time domain distortion of the signal arriving at our ears.

DRC Basic Steps

1. Measurement: you can only correct what you know
2. Target design: you have to define what you want
3. Inversion: the difference target minus measurement results in the correction (= inverse)
4. Convolution: the application of the resulting filters

Before we can actually create a solution, We need to better understand and model the problem domain. We need to better understand DSP.

What is Digital Signal Processing (DSP)?

"The world of science and engineering is filled with signals: images from remote space probes, voltages generated by the heart and brain, radar and sonar echoes, seismic vibrations, and countless other applications.

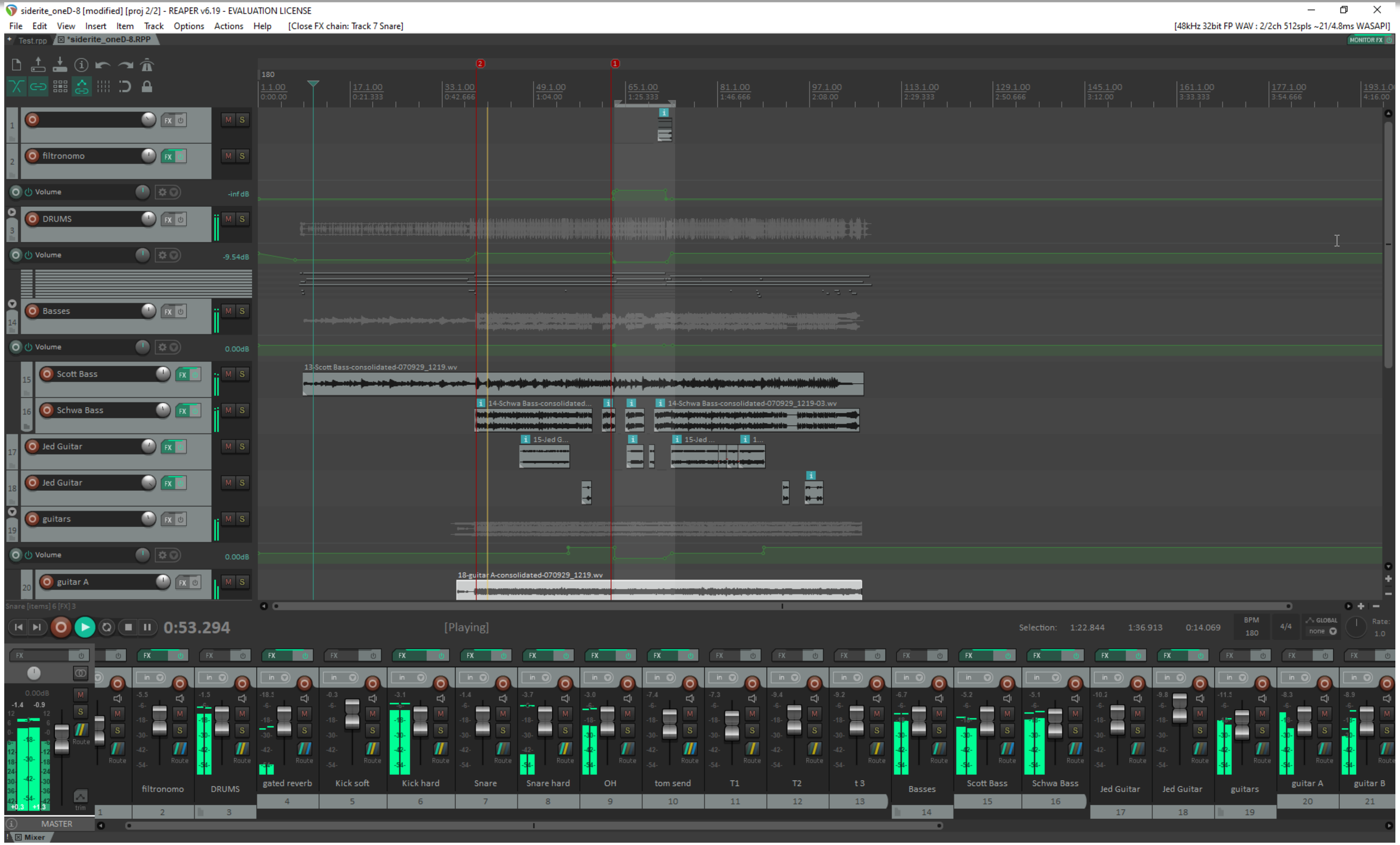
Digital Signal Processing is the science of using computers to understand these types of data. This includes a wide variety of goals: filtering, speech recognition, image enhancement, data compression, neural networks, and much more. **DSP is one of the most powerful technologies that will shape science and engineering in the twenty-first century.**

Suppose we attach an analog-to-digital converter to a computer, and then use it to acquire a chunk of real world data. DSP answers the question: What next?"

The Scientist & Engineer's Guide to Digital Signal Processing 1997

How DSP Revolutionized the Pro Audio Industry

Digital Audio Workstation (DAW) $60



https://www.reaper.fm/

UA 1176 Peak Limiter Electronic Device

The simple reality is that we have all heard the sound of this device since 1967 – that’s 50 years ago folks. In fact, if you are listening to mainstream music right now, it is likely that you are also listening to the sound of this device.

https://media.uaudio.com/assetlibrary/1/1/1176ln_manual.pdf

UA 1176 Peak Limiter in DSP Software

1176 Classic
Limiter Collection

~~$299.00~~ $179.00

You'll Save 40%

Add to Cart

The original Universal Audio 1176 was designed by UA founder Bill Putnam in 1967, and represented a major breakthrough in limiter technology. The first compressor featuring solid-state circuitry and ultra-fast 20 microsecond FET gain reduction, the 1176 is an easy-to-use “desert island” compressor that has lent its character and punch to some of the greatest recordings in history.

Upon its release in 2001, UA’s first 1176 plug-in single-handedly launched our UAD platform. **Extensive end-to-end circuit modeling** in 2013 captured even more sonic nuance. Now with updated graphics and additional controls, the 1176 Classic Limiter Collection continues its legacy of analog modeling excellence

Studer A-800 24 Track Recorder

Cost $60K in the late 80's
Weighs 700 lbs
Takes 8 hours to calibrate

Hardware → Software

DSP Transformation

Give it a listen!
https://youtu.be/gszni4TOATo

Studer® A800 Multichannel Tape Recorder

★★★★★

~~$349.00~~ $209.00
You'll Save 40%

Add to Cart

DSP Modelling – How is it Done?

There are two basic ways for a DSP designer to develop a processing module to mimic the sonic performance of a specific analogue or digital processor:

1. One is to pass a variety of static and changing signals through the device, measure the input-to-output characteristics for all front-panel settings, and then develop DSP (Digital Signal Processor) code that accurately emulates those changes.
2. The other is to examine the circuit diagram and model the various component blocks (using one of several commercially available programs), to generate a transfer function from input to output(s). That mathematical function can then be used to generate the DSP routines that emulate the device in question.

Most DSP developers combine both techniques, along with a lot of code refinements based on prior modelling experience, and some intensive listening sessions.

The Transfer Function

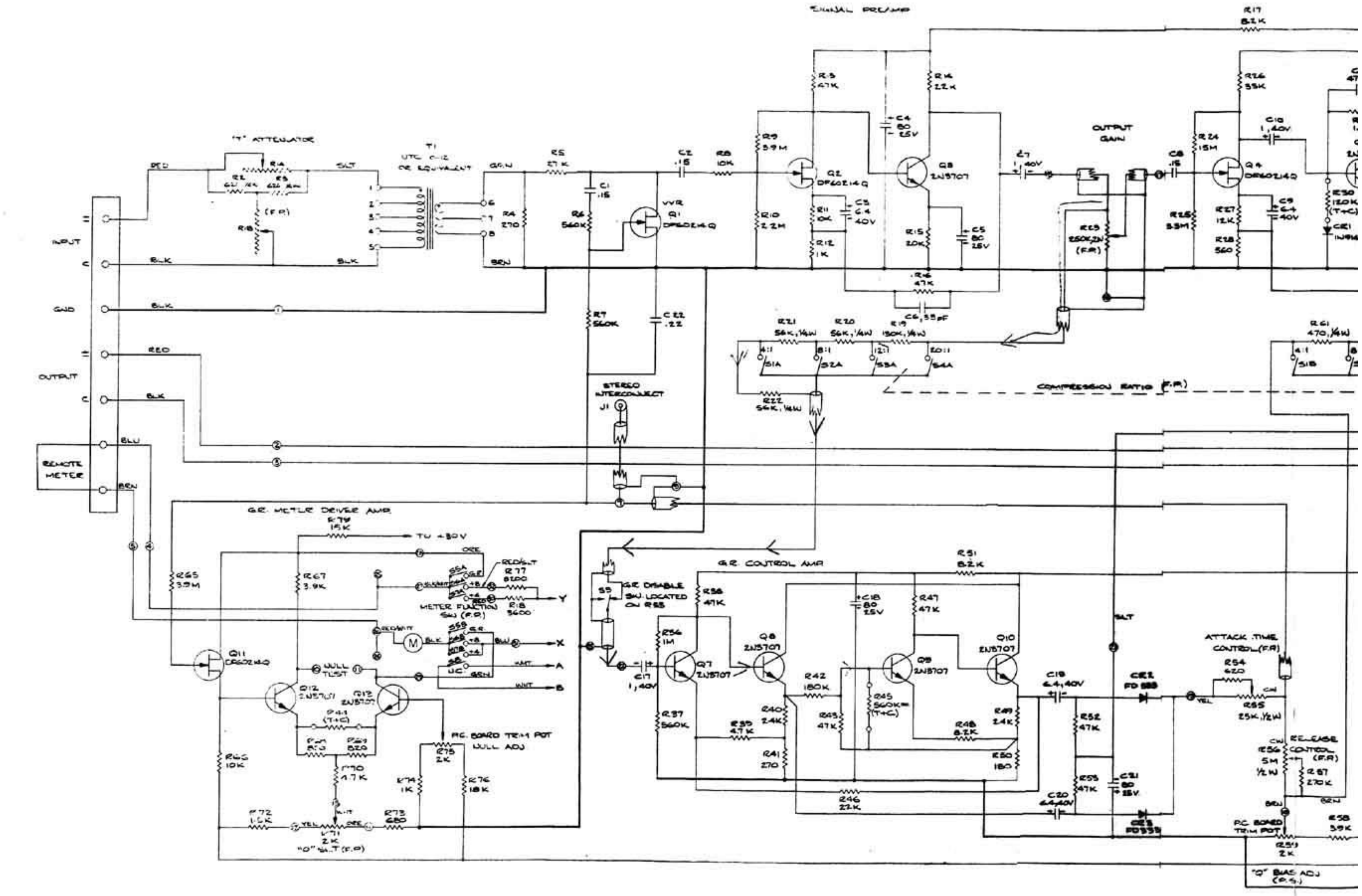
- At the heart of the modelling process is the development of DSP code that mimics the frequency response to time-variable signals, in response to adjustments via some form of graphical user interface — in other words, the on-screen knobs and switches.

- If a circuit diagram can be located for the device that is to be modelled, the first step is to use experience and/or a commercially available program — and usually a combination of both — to develop a multi-dimensional differential equation that relates output levels to variations in input levels and control parameters.

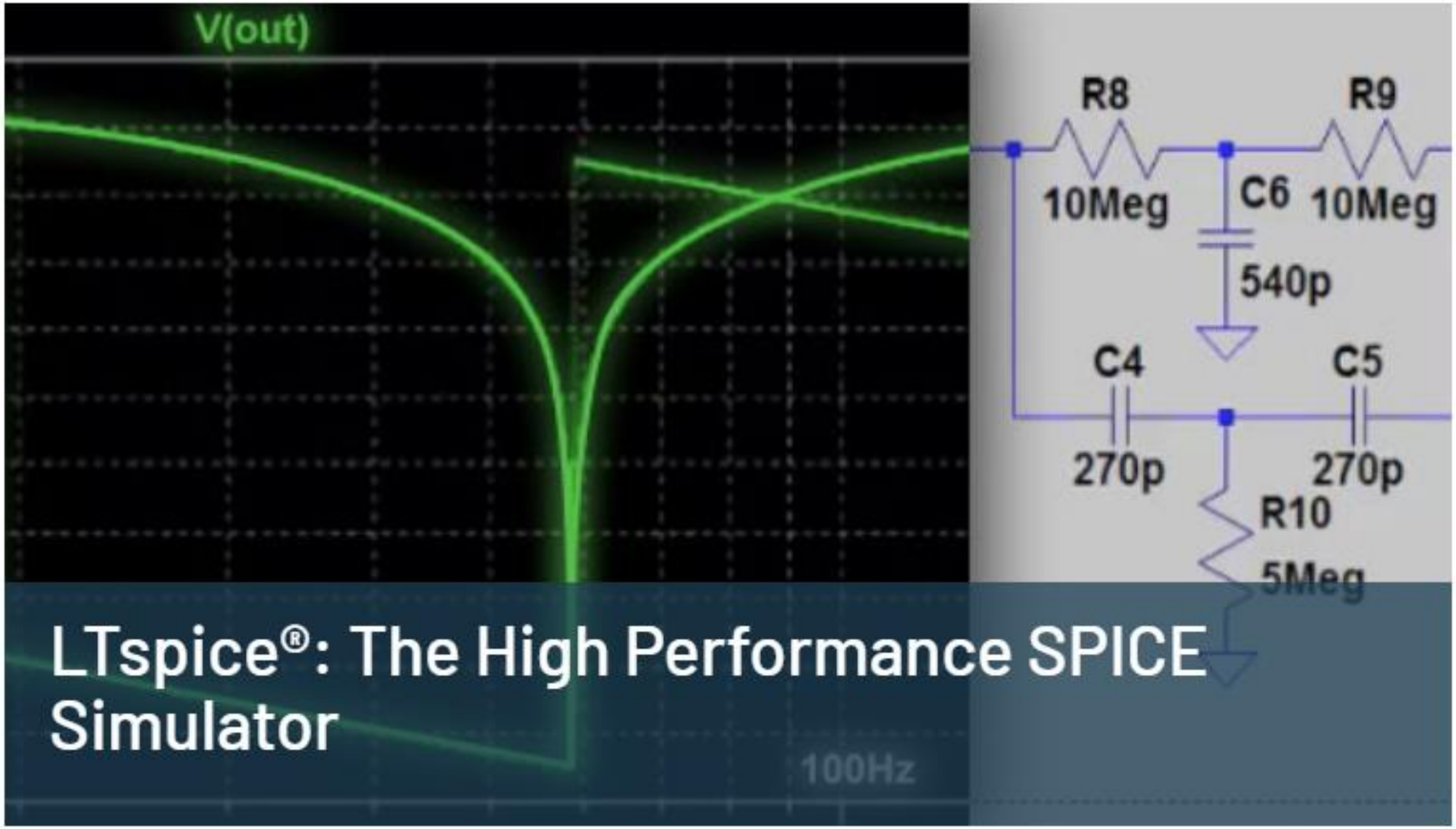
- That mathematical expression is the basic "[transfer function](https://docs.juce.com/master/classes.html)" for the device, which can then be modelled in DSP code.

Juce audio class library:
https://docs.juce.com/master/classes.html

UA 1176 Electronic Circuit Diagram



Example of Modelling Tools Used



https://www.analog.com/en/design-center/design-tools-and-calculators/ltspice-simulator.html

Modelling Loudspeakers in Rooms Using DSP

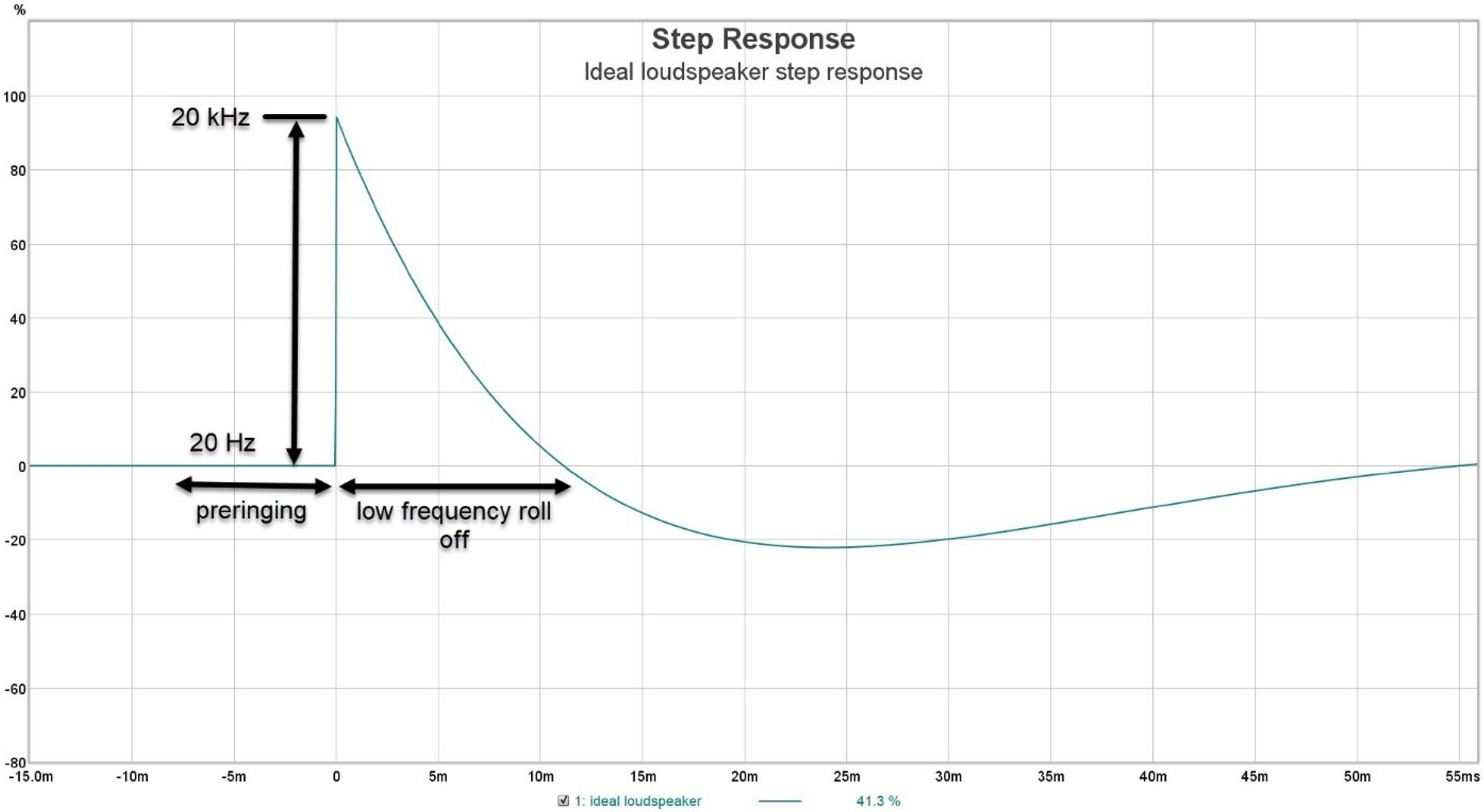
- The "ideal" loudspeaker can be modelled as a minimum phase system.
- A physical loudspeaker's impulse response can be measured.
- Rooms can be modelled and measured, but is more complicated as there are more variables at play. We will go through each one.
- A **transfer function** can be designed to restore the ideal loudspeakers minimum phase response to one's ears using a linear phase Finite Impulse Response (FIR) filter. The FIR filter contains both frequency and timing corrections that work independently.

Modelling the "Ideal" Loudspeaker

- 20 Hz to 20 kHz frequency response
  - flat frequency response +- 1 dB tolerance (we can hear ~1 dB eq difference)
  - roll off after 20 Hz as speakers typically do not go down to DC or 0 Hz.
  - "ideal" loudspeaker is a minimum phase system
- No crossover network – no potential phase distortions
- Ideal directivity response (see Earl Geddes paper on Directivity in Loudspeaker Systems)
- No room – i.e. anechoic response
- Lets open up a FIR filter designer tool to model the ideal loudspeaker

The “Ideal” Loudspeaker – Frequency Response

The “Ideal” Loudspeaker - Timing Response



Standard Method for Measuring Loudspeakers

"This free standard describes an improved method for measuring and reporting the performance of a loudspeaker in a manner that should help consumers better understand the performance of the loudspeaker and convey a reasonably good representation of how it may sound in a room based on its off-axis response and how this response affects the consumer's experience."

The Estimated In-Room Response shall be presented using the method described in Section 5.3.

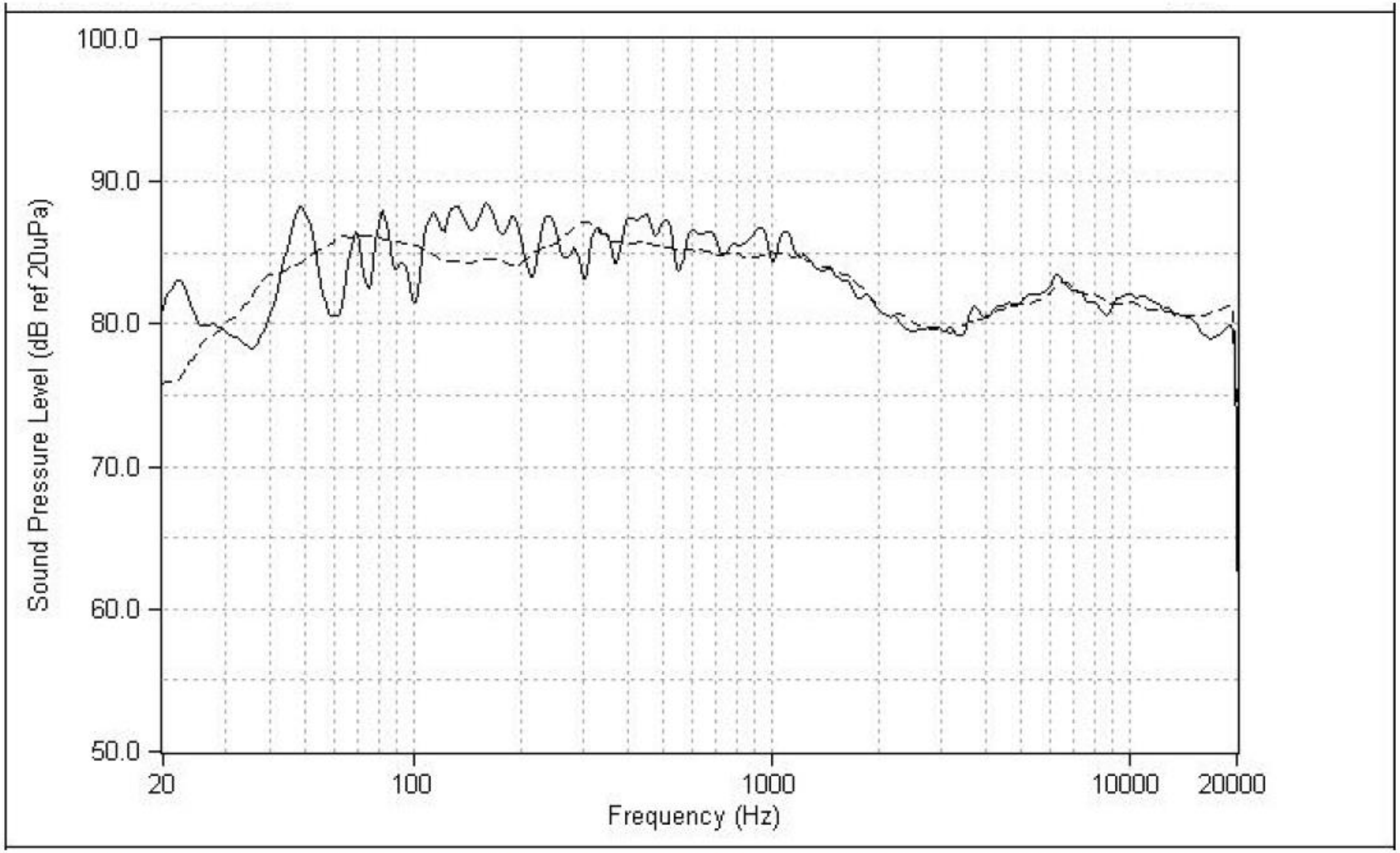


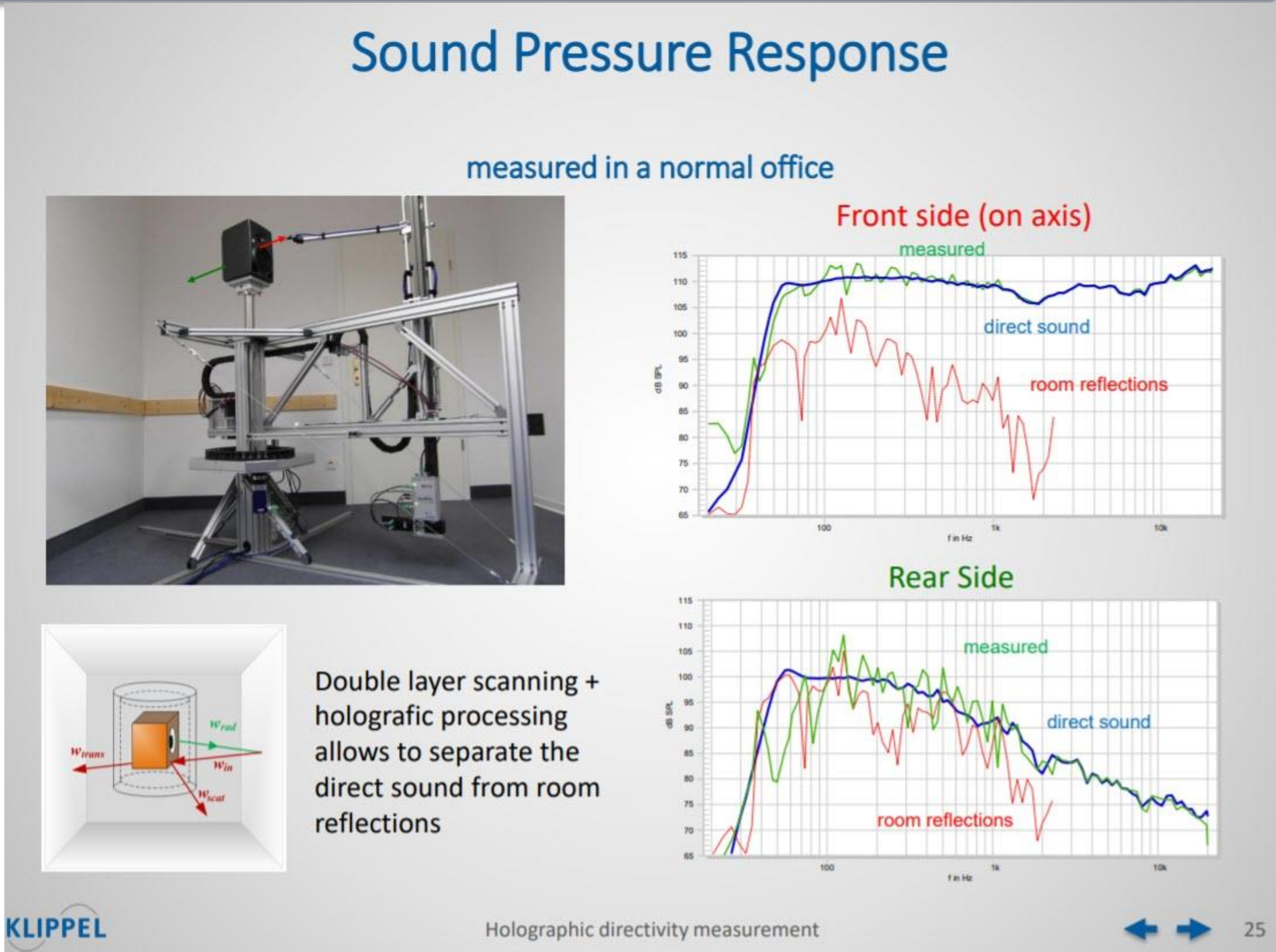
Figure 11: In-room response of loudspeaker B. Solid: predicted from anechoic data. Dotted: average response in a real room, with a ¼-inch microphone aimed at the ceiling, random-incidence amplitude response corrected to be flat. Note that the predicted curve cannot anticipate the effects of room modes at frequencies below the transition/Schroeder frequency around 300 Hz to 400 Hz.

Klippel Nearfield Scanner JBL M2 DUT

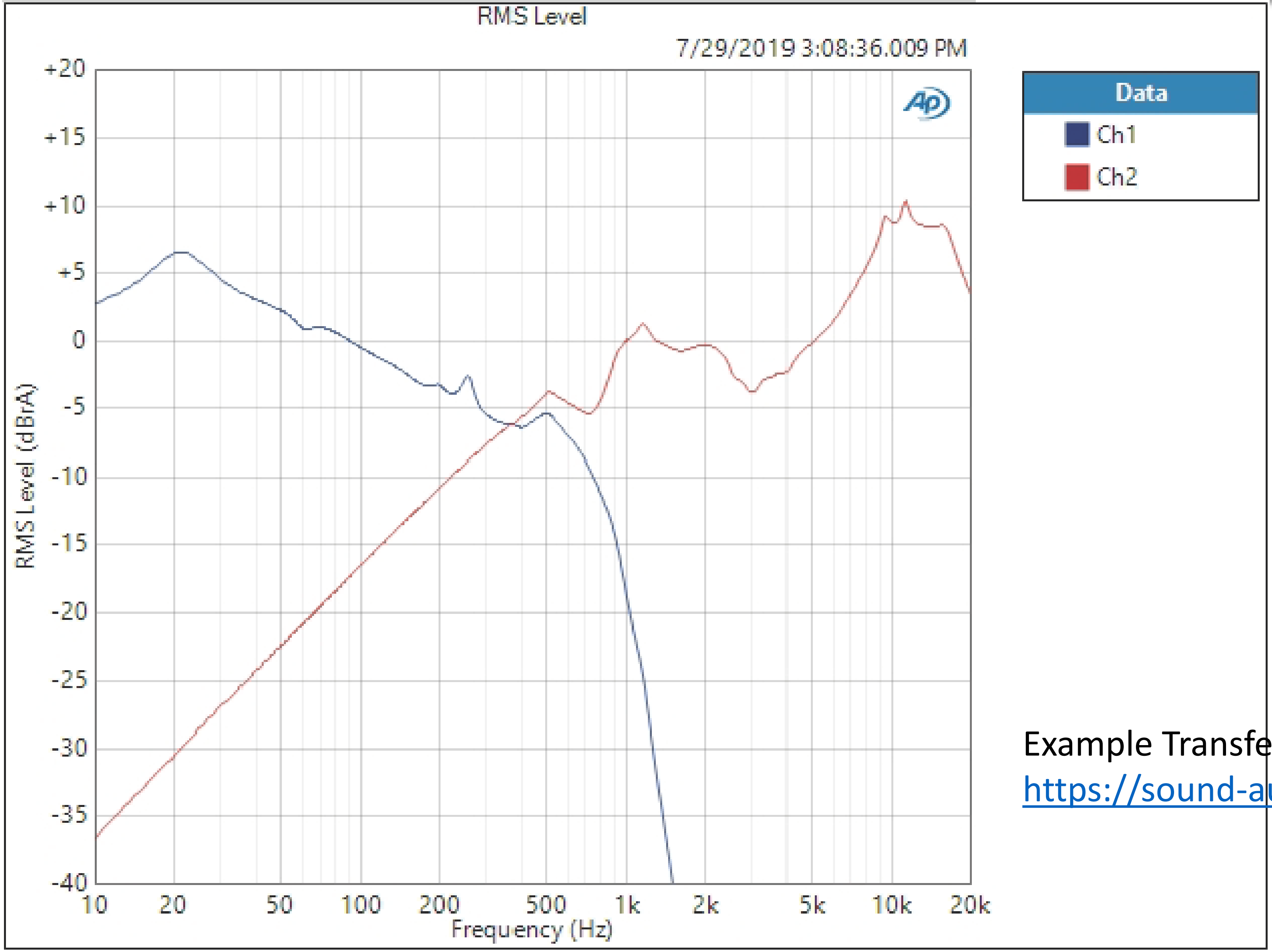
Klippel Nearfield Scanner using DSP

Standard method for measuring loudspeakers: “window out” low frequency room reflections

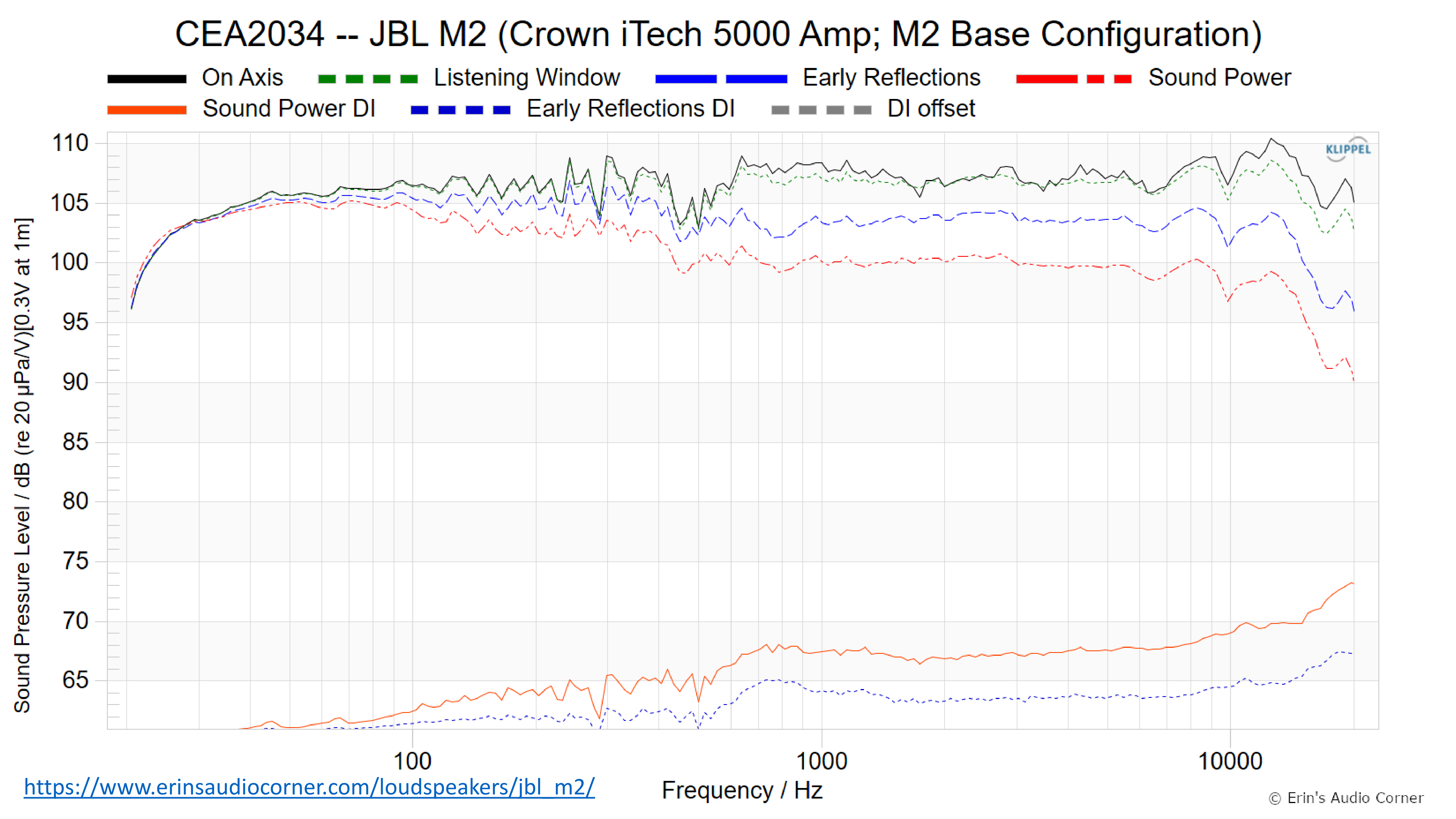
For DRC: let in the low frequency room reflections and “window out” mid and HF room reflections. Linearize the direct sound of the loudspeaker

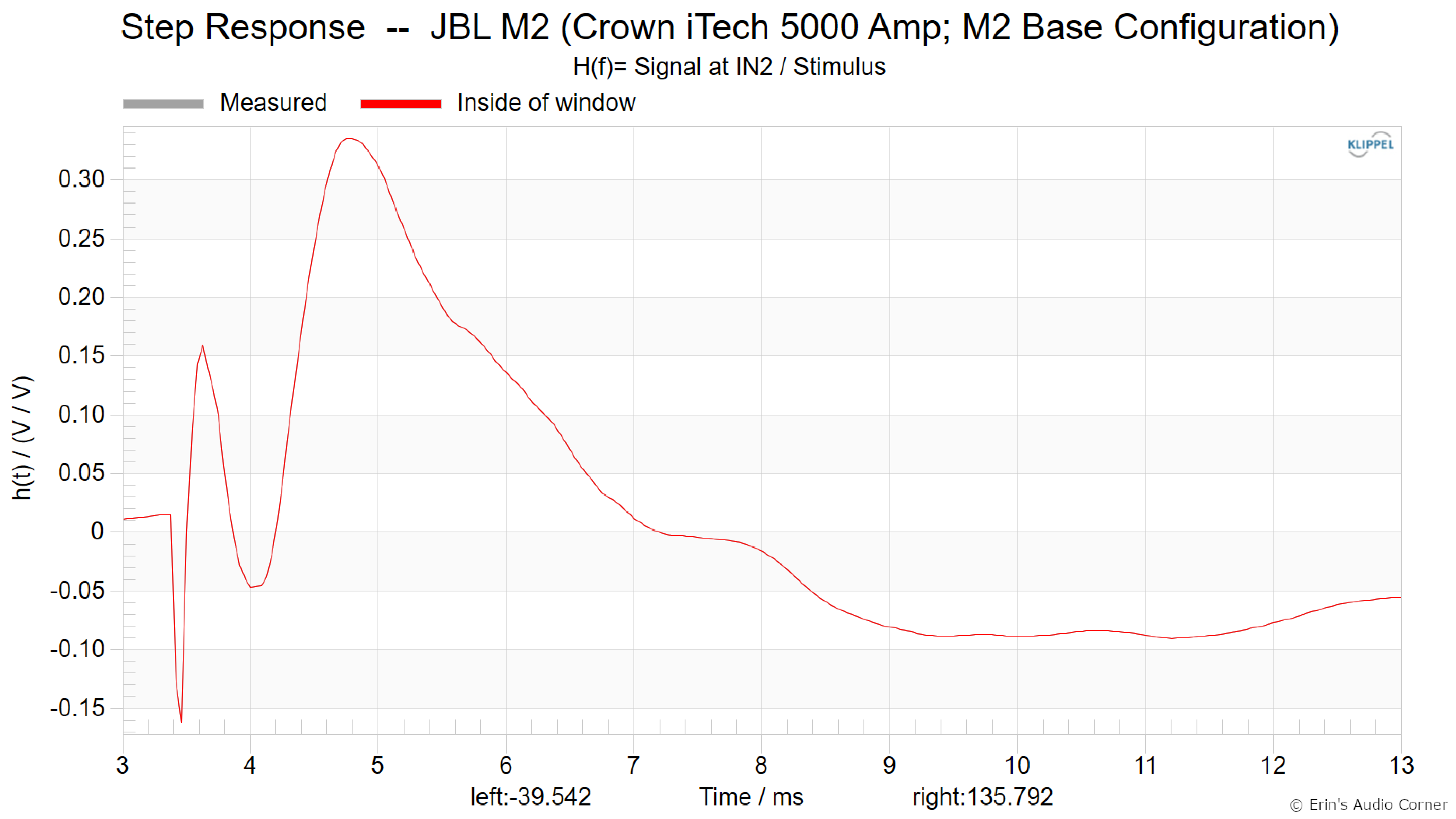


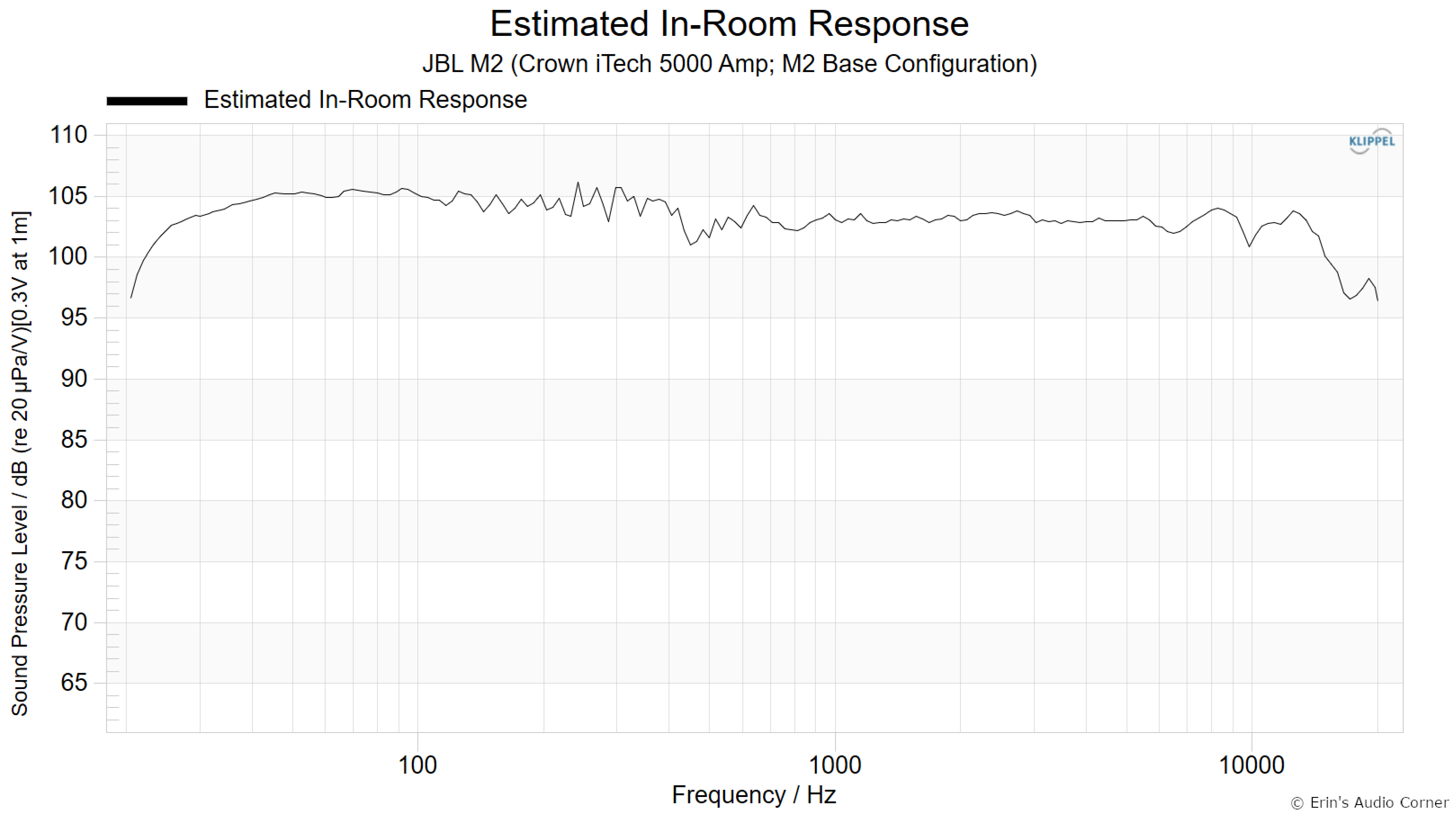
JBL M2 Loudspeaker DSP EQ Transfer Function



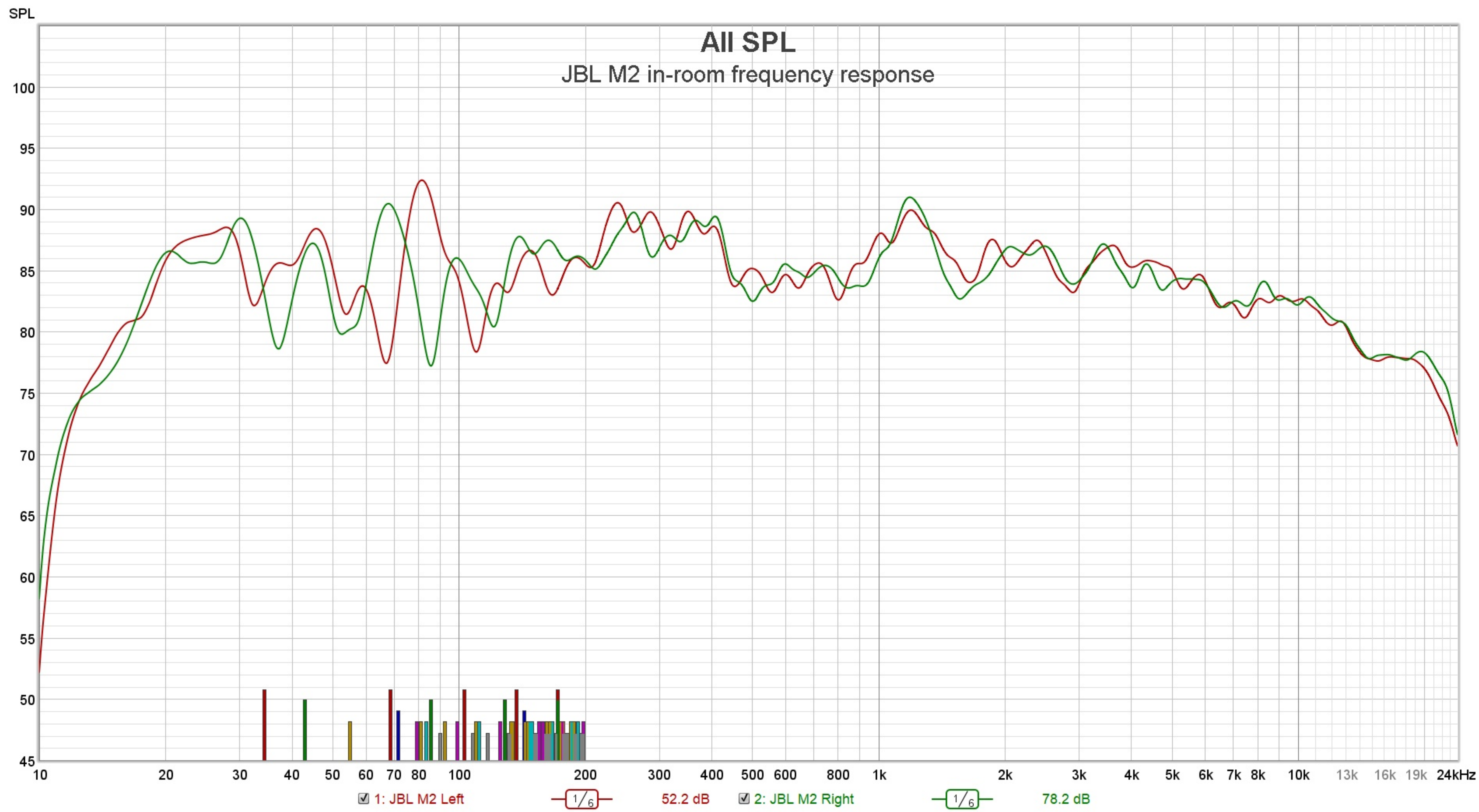
Example Transfer function
https://sound-au.com/project173.htm


CEA2034 -- JBL M2 (Crown iTech 5000 Amp; M2 Base Configuration)
On Axis
Listening Window
Early Reflections
Sound Power
Sound Power DI
Early Reflections DI
DI offset
KLIPPEL
Sound Pressure Level / dB (re 20 µPa/V)[0.3V at 1m]
110
105
100
95
90
85
80
75
70
65
100
1000
10000
Frequency / Hz
https://www.erinsaudiocorner.com/loudspeakers/jbl_m2/
© Erin's Audio Corner


Step Response -- JBL M2 (Crown iTech 5000 Amp; M2 Base Configuration)
H(f)= Signal at IN2 / Stimulus
Measured
Inside of window
KLIPPEL
h(t) / (V / V)
0.30
0.25
0.20
0.15
0.10
0.05
0
-0.05
-0.10
-0.15
3
4
5
6
7
8
9
10
11
12
13
left:-39.542
Time / ms
right:135.792
© Erin's Audio Corner

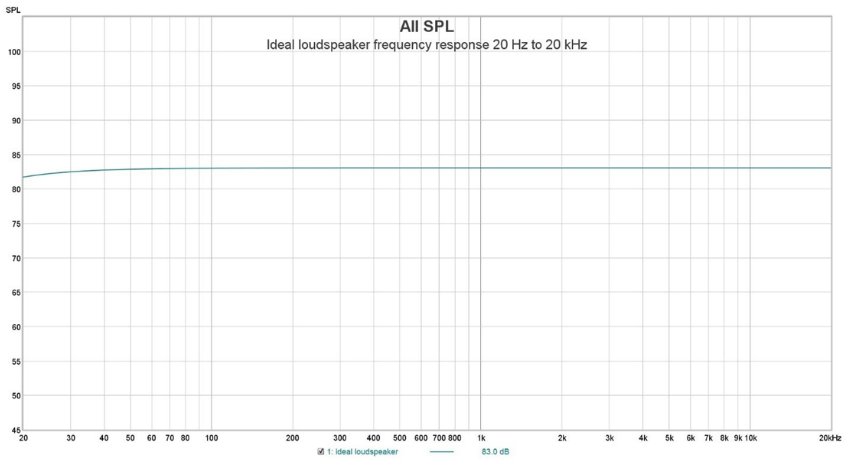


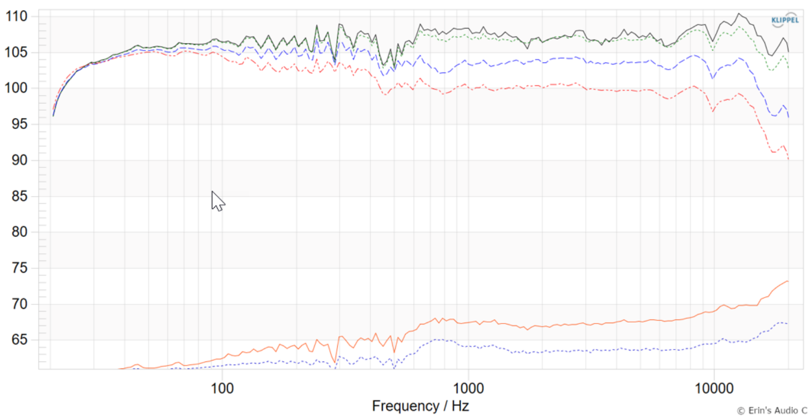
Estimated In-Room Response
JBL M2 (Crown iTech 5000 Amp; M2 Base Configuration)
Estimated In-Room Response
KLIPPEL
110
105
100
95
90
85
80
75
70
65
Sound Pressure Level / dB (re 20 µPa/V)[0.3V at 1m]
100
1000
10000
Frequency / Hz
© Erin's Audio Corner

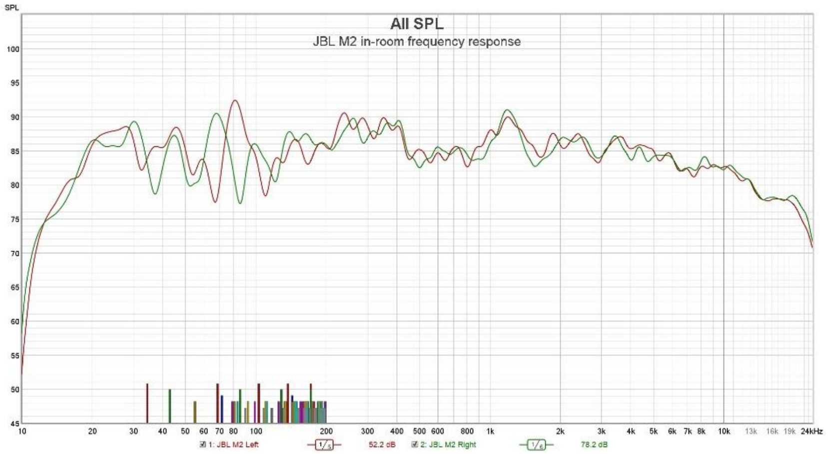

All SPL
JBL M2 in-room frequency response
SPL
100
95
90
85
80
75
70
65
60
55
50
45
10
20
30
40
50
60
70
80
100
200
300
400
500
600
800
1k
2k
3k
4k
5k
6k
7k
8k
10k
13k
16k
19k
24kHz
1: JBL M2 Left
1/6
52.2 dB
2: JBL M2 Right
1/6
78.2 dB

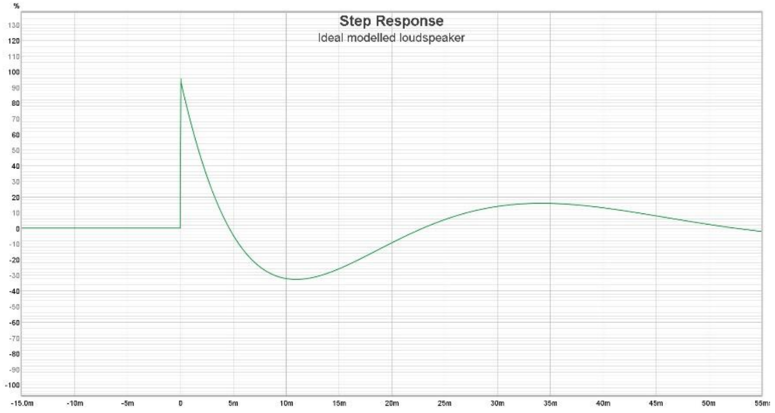

Step Response
JBL M2 in-room step response
%
130
120
110
100
90
80
70
60
50
40
30
20
10
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
-100
-1.04m
-500u
0
500u
1.0m
1.5m
2.0m
2.5m
3.0m
3.5m
4.0m
4.5m
5.0m
5.5m
6.0m
6.5m
7.0m
7.5m
8.0m
8.5m
9.0m
9.5m
10.0ms
1: JBL M2 Left
-21.9 %
2: JBL M2 Right
-31.9 %

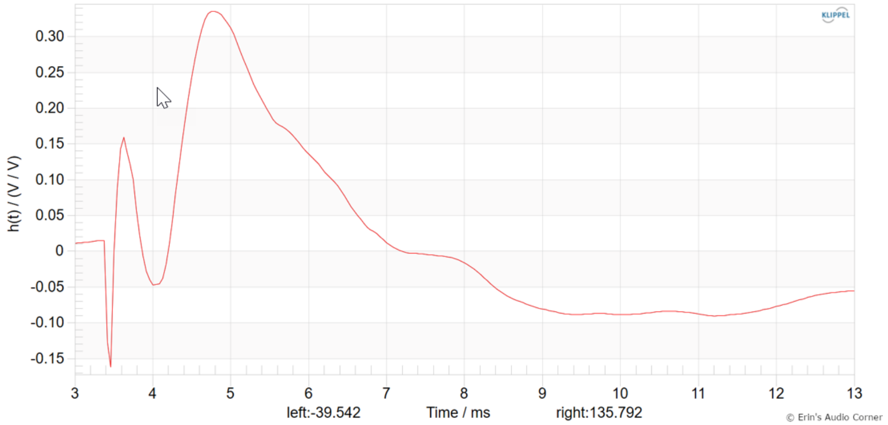
Ideal Model vs Measured vs Loudspeaker In-Room



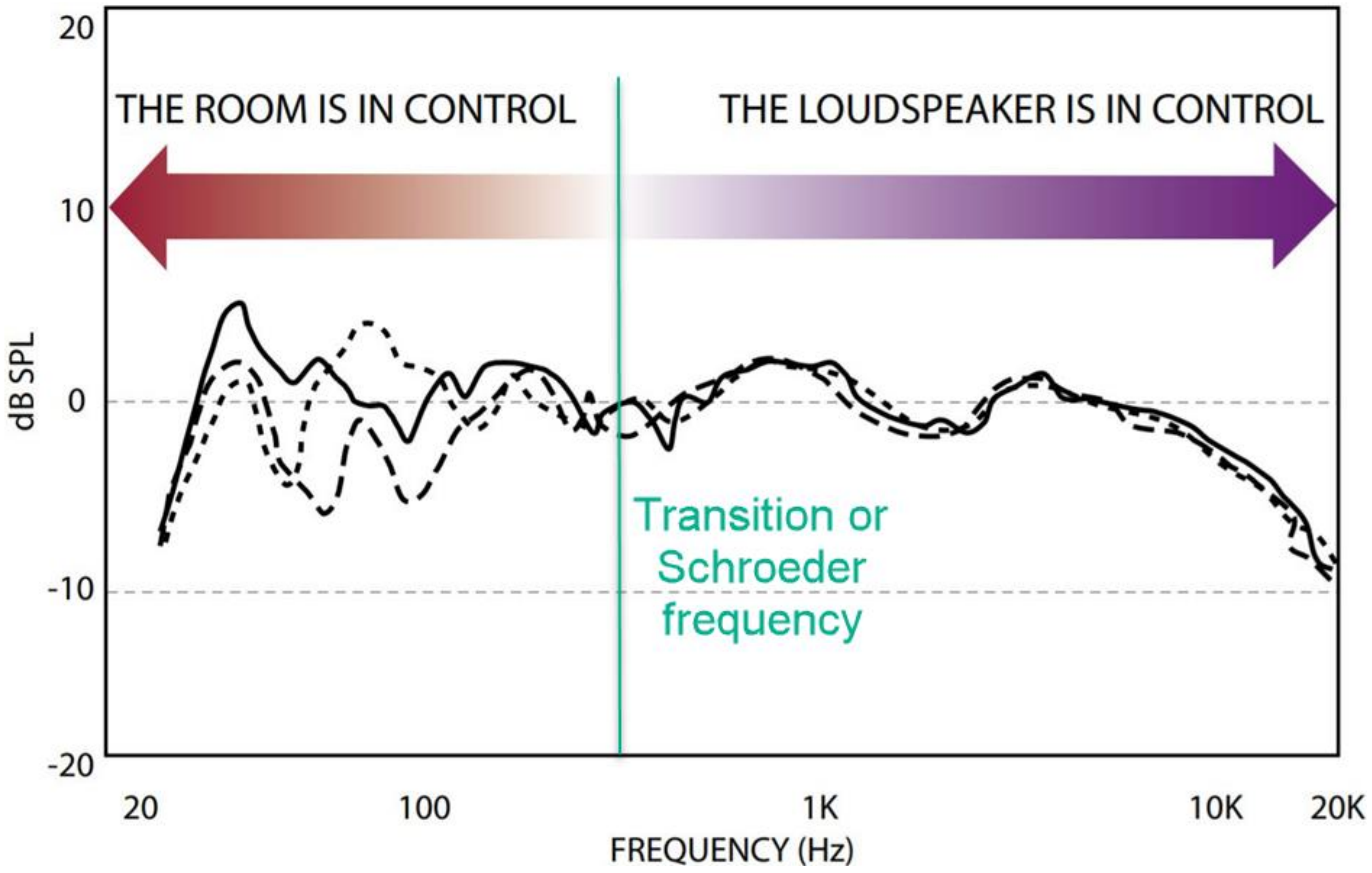








At Low Frequencies the Room is in Control



As Floyd Toole says, “In the investigation of many rooms over the years, I would estimate that something like **80% have serious bass coloration.**”

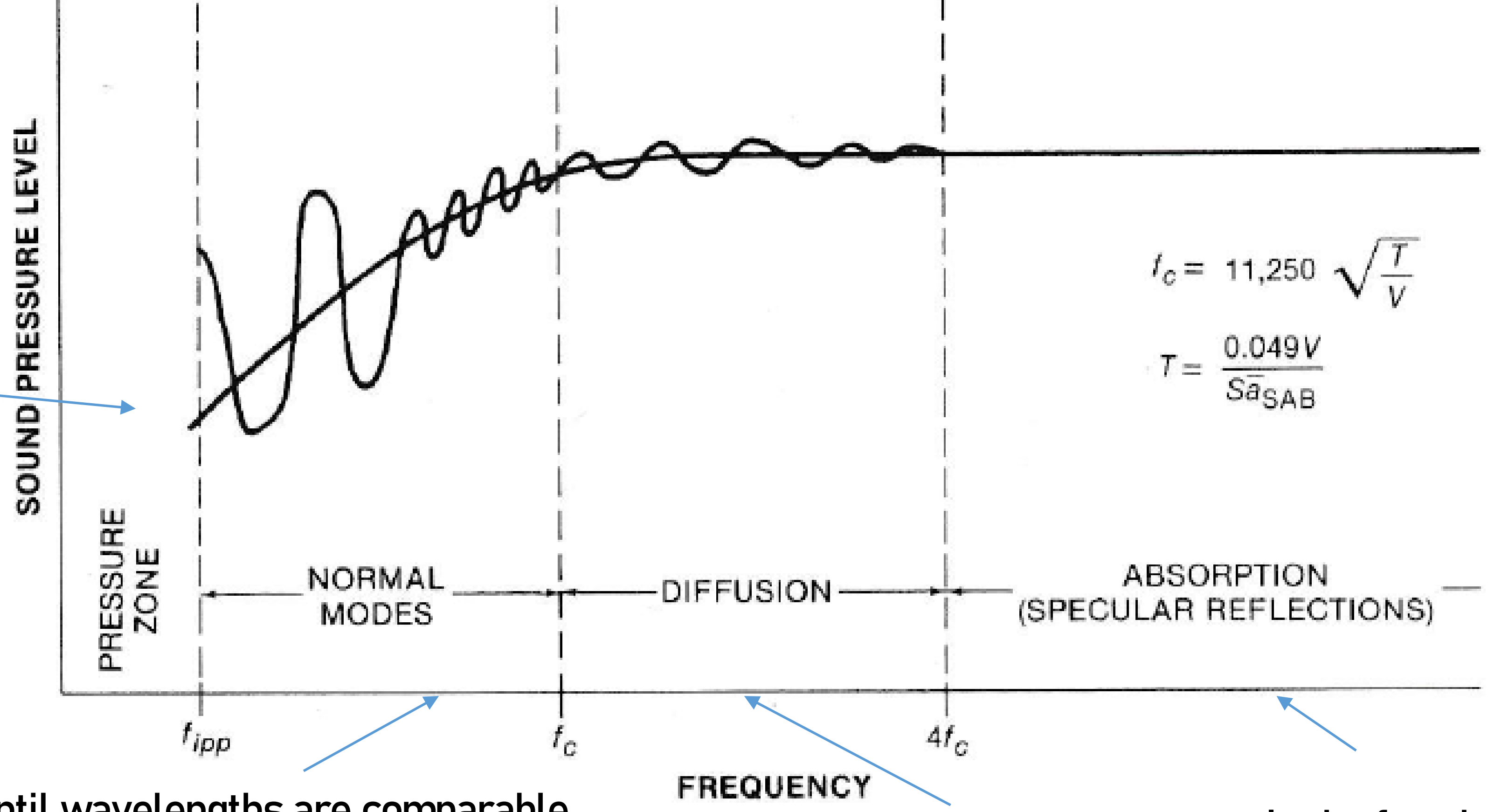
Further, Floyd’s research shows that bass subjectively accounts for 30% of how we judge speakers sound quality.”

“ANY loudspeaker can sound better after room EQ, so long as it competently addresses the bass frequencies - this is not a guarantee, but really is not difficult for at least the prime listener.”

Controllers of steady state acoustic room response

The first zone is below the frequency that has a wavelength of twice the longest length of the room. In this zone, sound behaves very much like changes in static air pressure.

Courtesy Bolt, Beranek and Newman. Sound System Engineering, Don and Carolyn Davis.



Above that zone, until wavelengths are comparable to the dimensions of the room, room resonances / standing waves dominate. This transition frequency is popularly known as the Schroeder frequency.

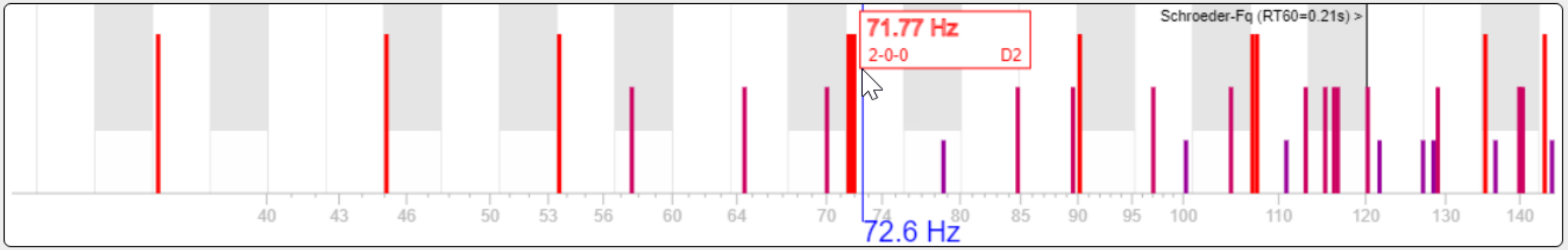
The third region which extends approximately two octaves is a transition to the fourth zone.

In the fourth zone, sounds behave like rays of light bouncing around the room.

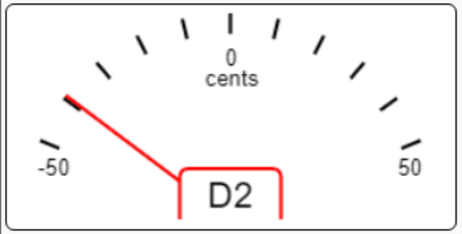
Room Modes are Based on Room Dimensions

Room dimensions:

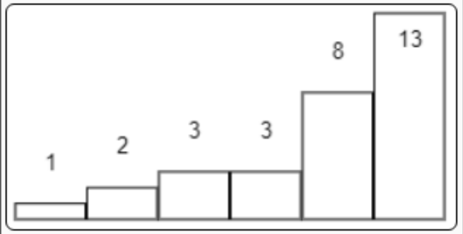
length: < 15.68 >
width: < 12.48 >
height: < 10.5 >
cm feet



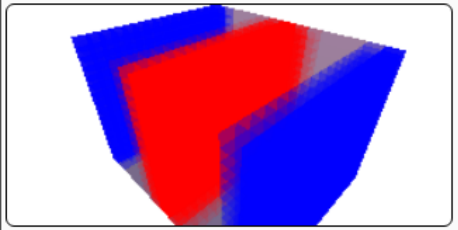
Tune of the selected mode:



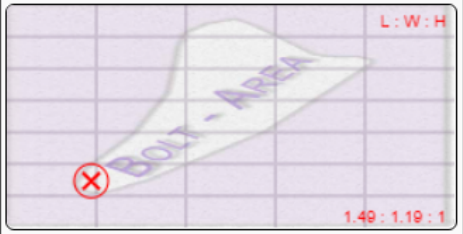
Bonello (modes per third):



Room 3D (2-0-0)



Bolt-area:



Additional data about the room:

Volume: ***2054.73 cft***

Surface: ***982.73 sq ft***

RT60: ***0.21*** ***sec*** EBU listening room *EBU Tech. 3276 - 2nd edition, May 1998, p.6*

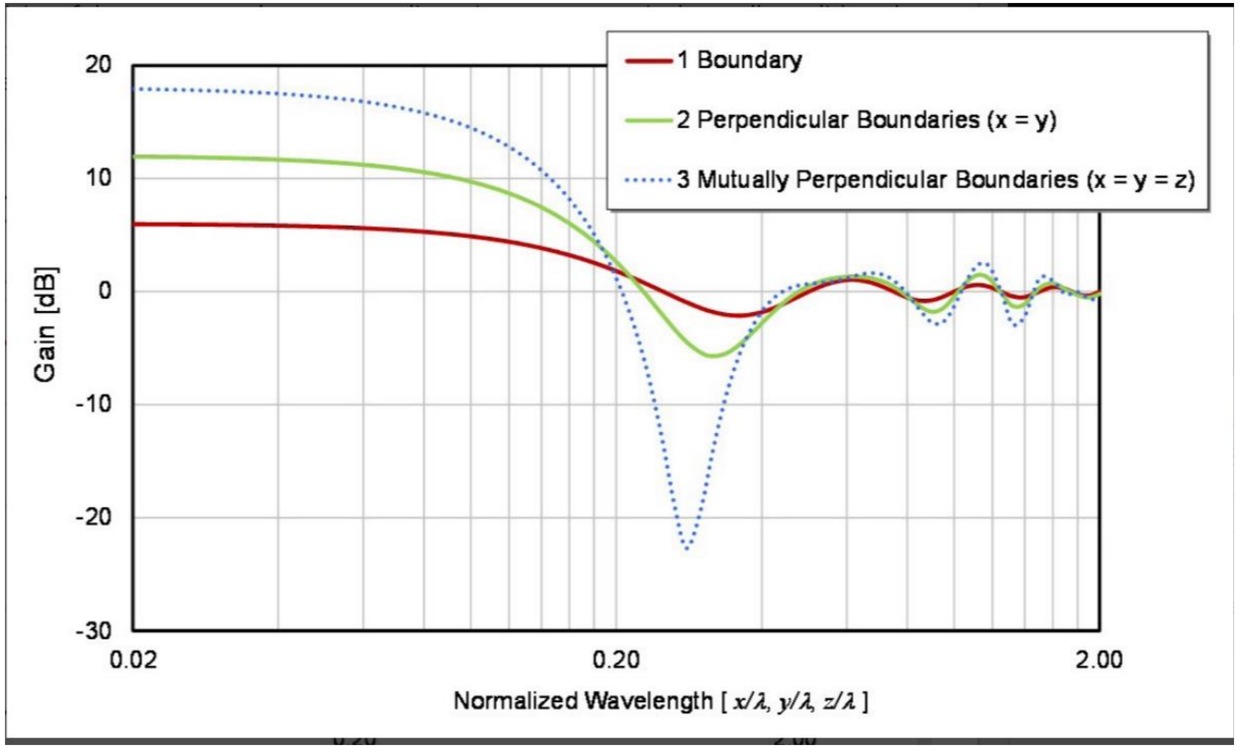
Schroederfrequency: ***120 Hz***

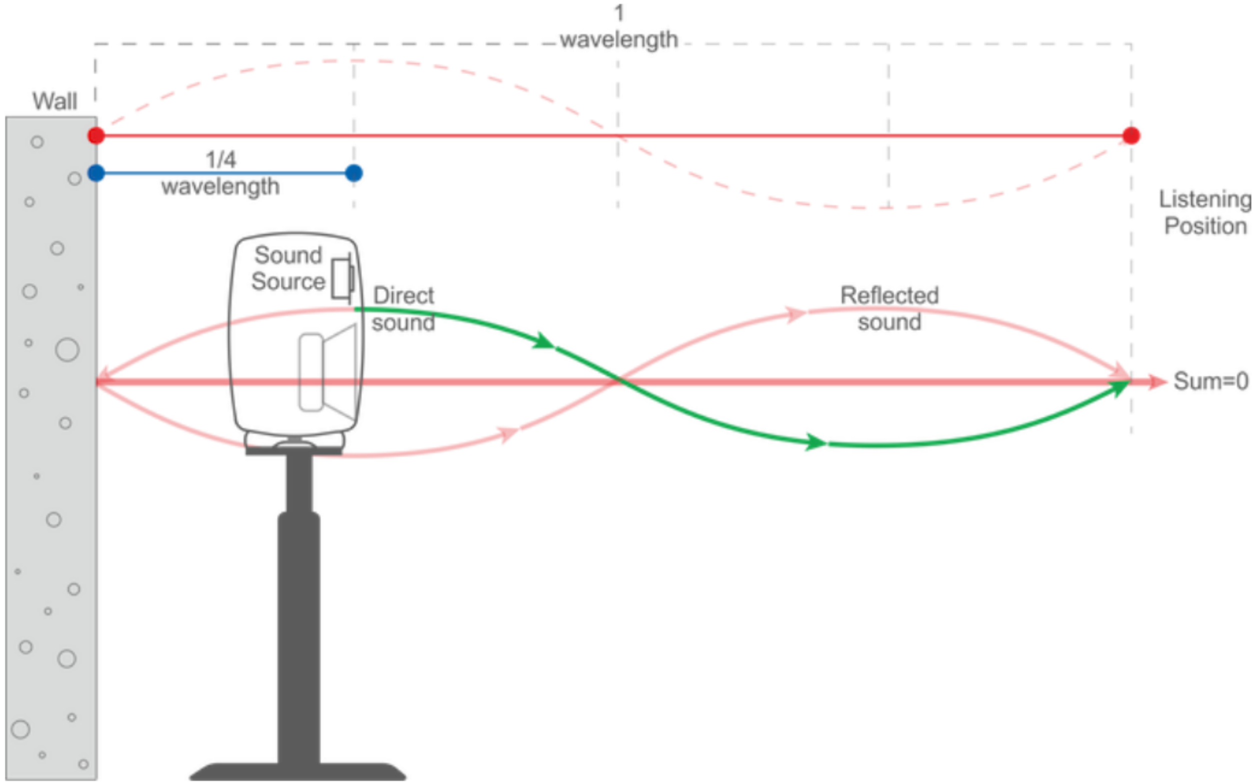
Critical distance: ***3.11 ft***

https://amcoustics.com/tools/amroc

Frequency in Hz.	Wavelength	quarter wavelength	
20	56.50	14.13	feet
25	45.2	11.3	feet
31.5	35.87	8.97	feet
40	28.25	7.06	feet
50	22.6	5.65	feet
63	17.94	4.48	feet
80	14.13	3.53	feet
100	11.3	2.83	feet
125	9.04	2.26	feet
160	7.06	1.77	feet
200	5.45	1.41	feet
250	4.52	1.13	feet
320	3.53	0.88	feet
400	2.83	0.71	feet
500	27.12	6.78	inches
640	21.19	5.3	inches
800	16.95	4.24	inches
1000	13.56	3.39	inches
1280	10.59	2.65	inches
1600	8.48	2.12	inches
2000	6.78	1.7	inches
2560	5.3	1.32	inches
3200	4.24	1.06	inches
4000	3.39	0.85	inches
5120	2.65	0.66	inches
6400	2.12	0.53	inches
8000	1.7	0.42	inches
10240	1.32	0.33	inches
12800	1.06	0.26	inches
16000	0.85	0.21	inches
20480	0.66	0.17	inches

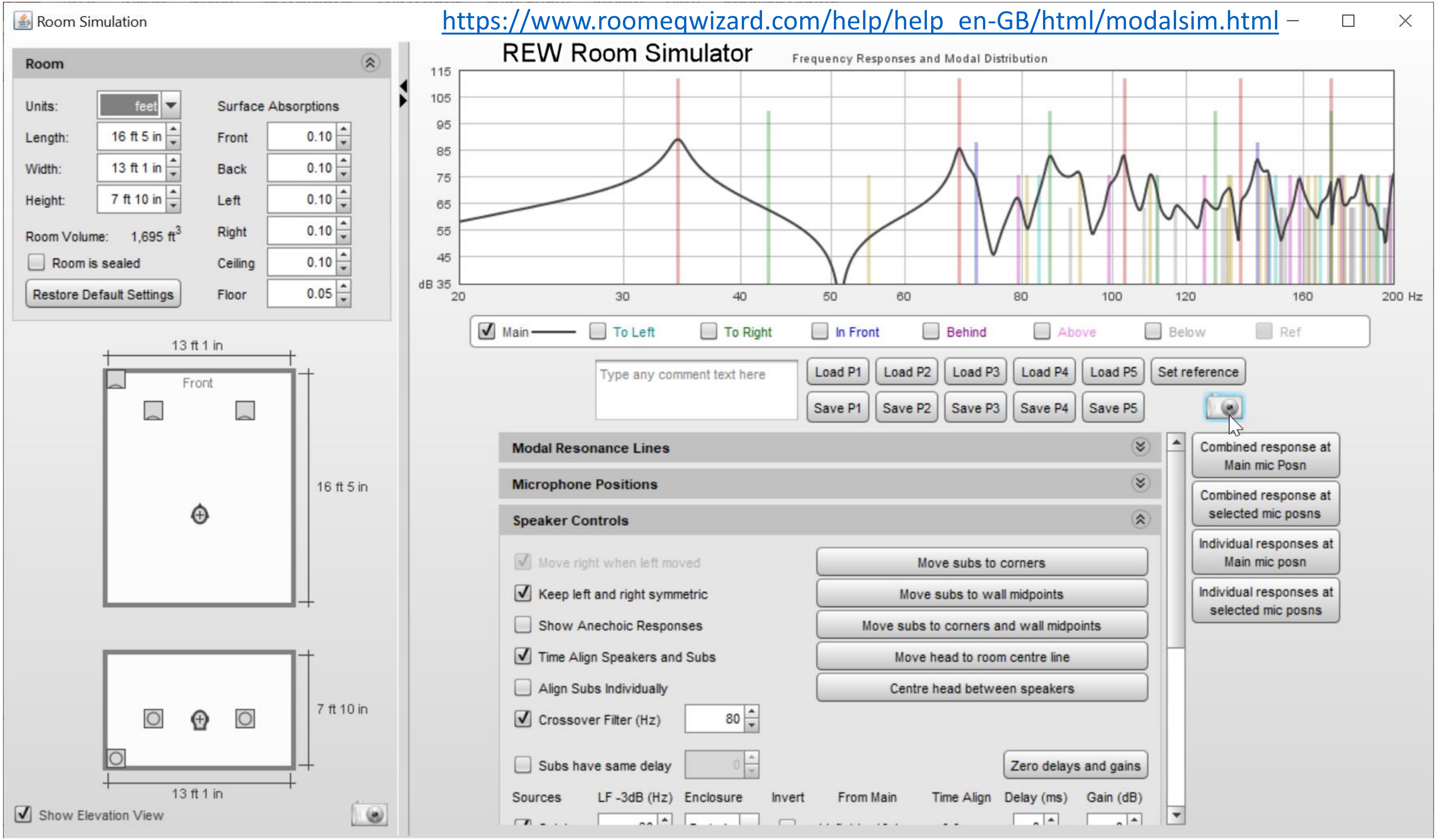
Speaker Boundary Interference Response



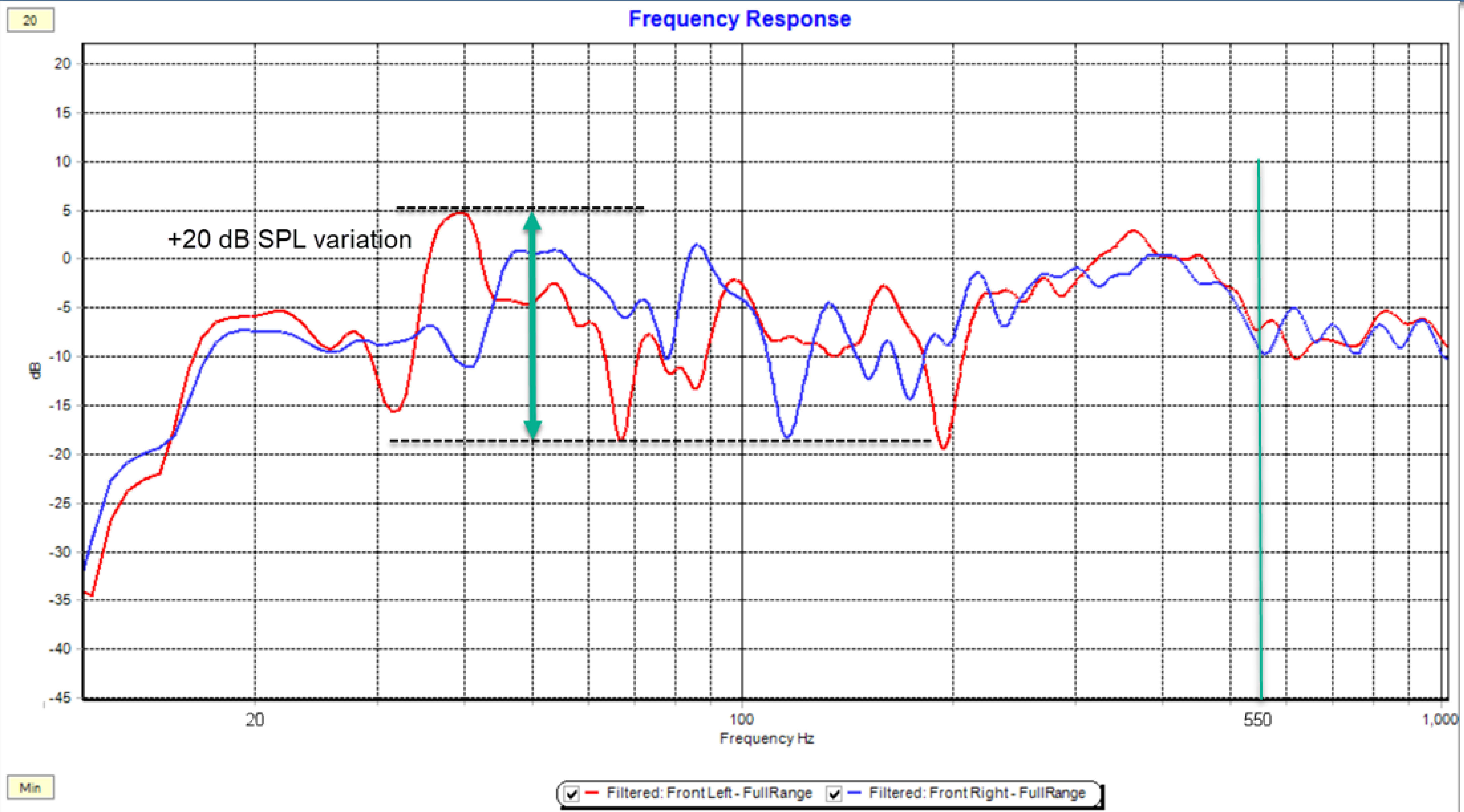


http://tripp.com.au/sbir.htm

https://www.roomeqwizard.com/help/help_en-GB/html/modalsim.html



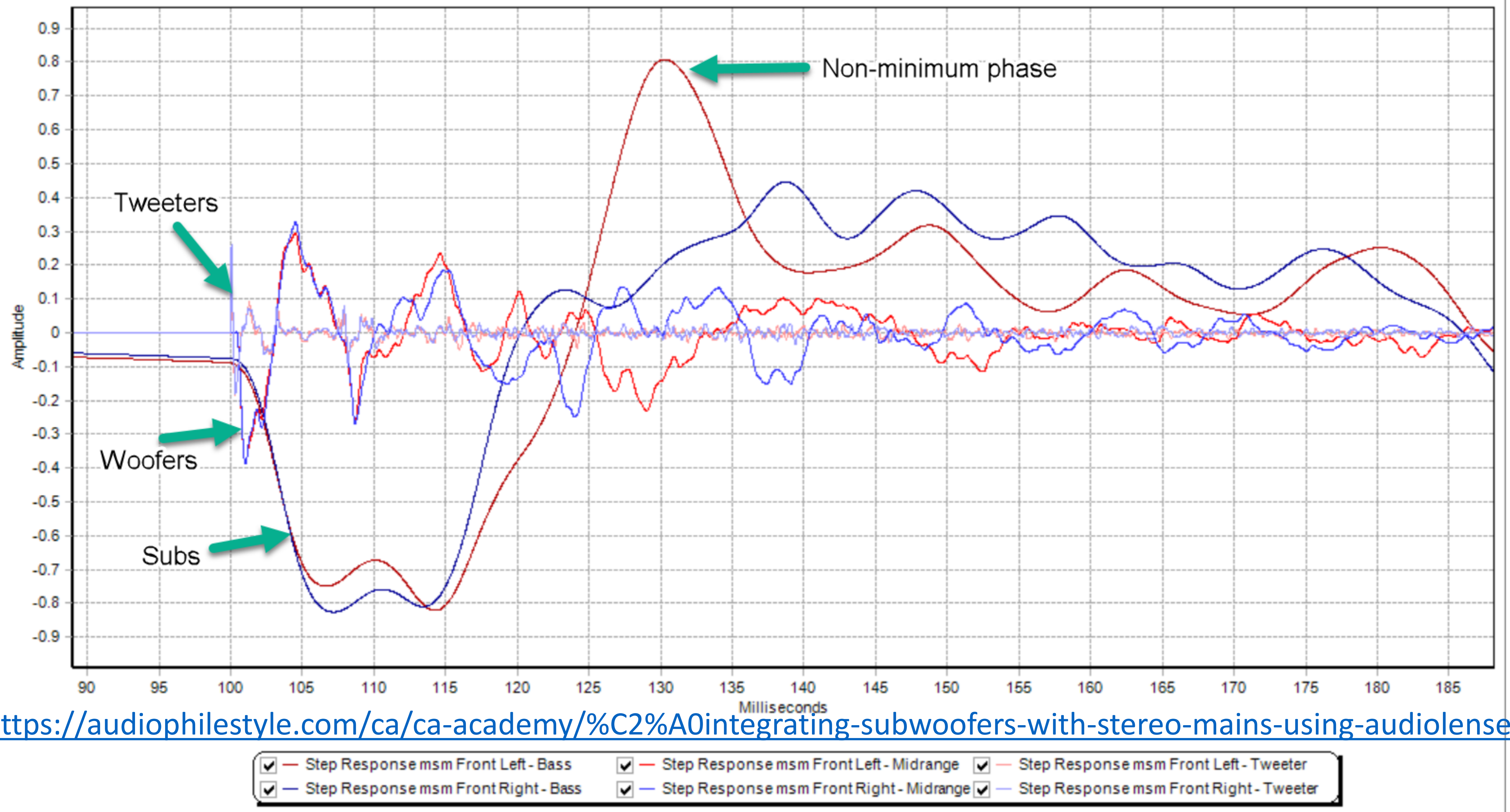
Loudspeaker in Room – Low Frequencies



Minimum Phase in Room Acoustics

- Minimum phase systems can be inverted, which means that a filter can be designed that, if applied to the system, would produce a flat response *and* correct the phase response at the same time.
- Most rooms have low frequency regions that are **not minimum phase**
- A simple example of something that renders a response non-minimum phase are reflections that are as large or larger than the direct signal (reflections along paths that are different but the same length can combine to produce higher levels).
- Another example are the axial modes in the room – we will see an example of that later.
- Check out John Mulcahy's, author of REW, topic on minimum phase.

Example of Non-Minimum Phase Behavior



https://audiophilestyle.com/ca/ca-academy/%C2%A0integrating-subwoofers-with-stereo-mains-using-audiolense-r712/

AES PNW Section Meeting Report

Acoustic and Psychoacoustic Issues in Room Correction

with James (JJ) Johnston & Serge Smirnov

James (JJ) Johnston (left) and Serge Smirnov (right) discuss room correction issues

[Download Powerpoint Presentation (421.9K PPT file)]

Photo by Gary Louie

AES papers from JJ:

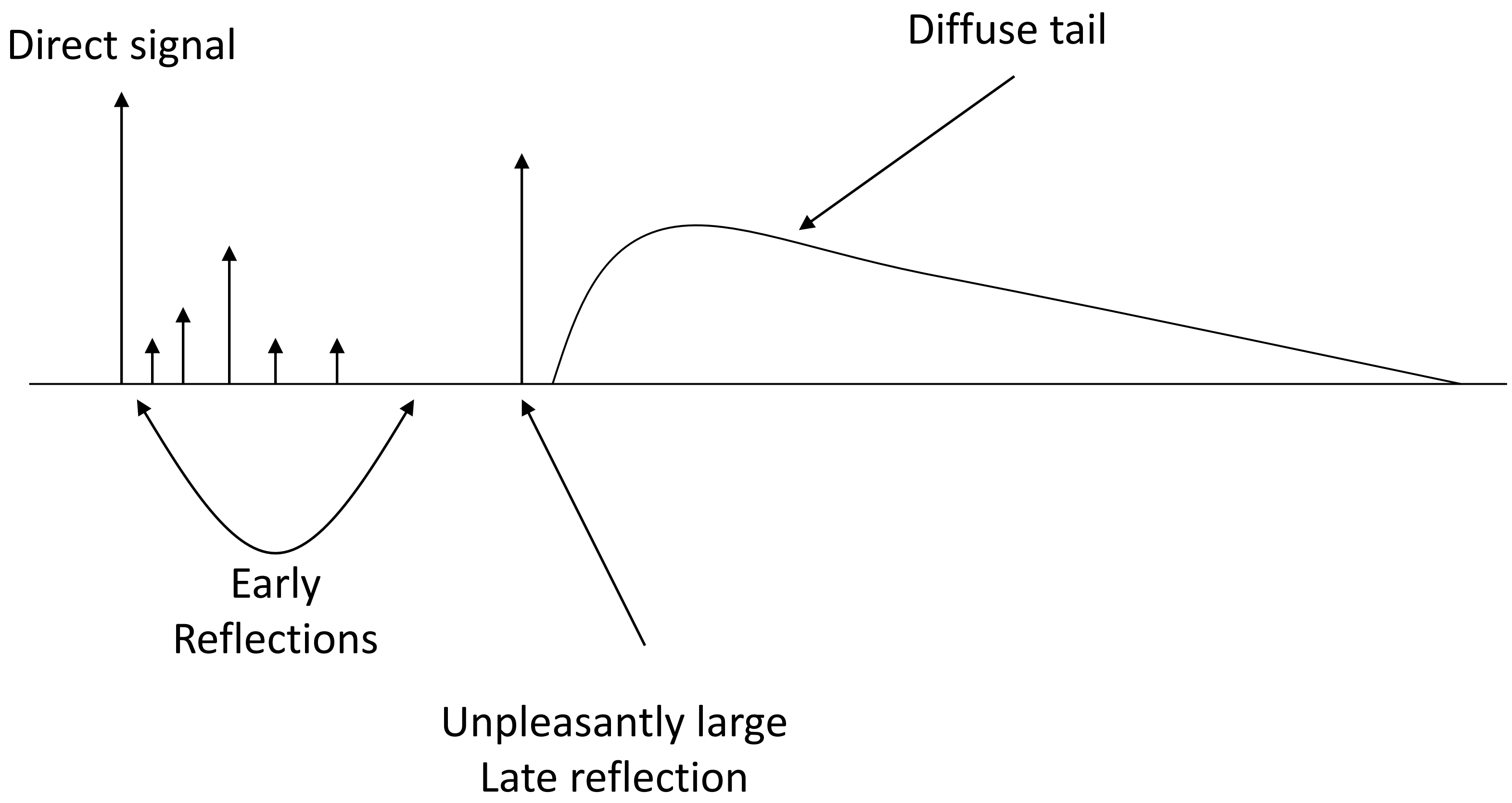
- A Low Complexity Perceptually Tuned Room Correction System.
- Beyond Coding: Reproduction of Direct and Diffuse Sounds in Multiple Environments
- DTS Multi-Channel Audio Playback System: Characterization and Correction

Library of open access papers from JJ:

- https://www.aes-media.org/sections/pnw/ppt/jj/
- See section on "Hearing and Psychoacoustics"

http://www.aes-media.org/sections/pnw/pnwrecaps/2008/jj_jan08/

Acoustics – What Does a Room Do?



Early reflections are those more or less under the 10 msec mark.
“late” (specular) reflections create a problem with perception.

JJ's Summary of Acoustic/Psychoacoustic Issues

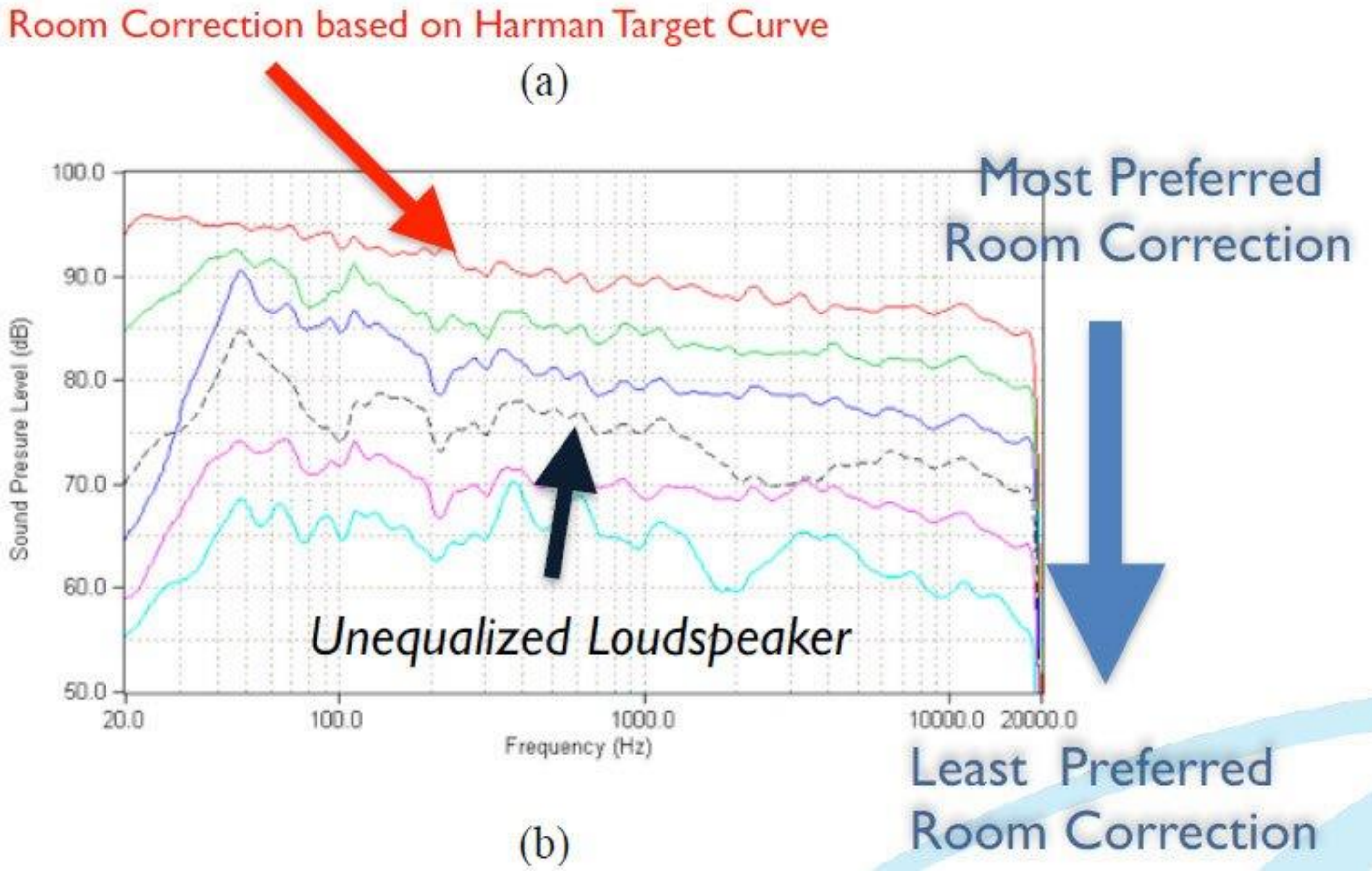
- In a room high frequencies decay faster than low frequencies.
- For loudspeakers in rooms the directivity at low frequencies is omnidirectional and narrows as the frequency goes up.
- This is why when we measure an in-room response, we get a tilted response with more energy at the low end than at the top end.
- The frequency response of the direct sound will be different than the measured response of the direct sound plus the rooms reflections.
- Room nulls – adding more energy into a null will do nothing but make it sound worse.
- Our ears are more sensitive to peaks than narrow band dips in frequency response. Our ears follows the spectral envelope.

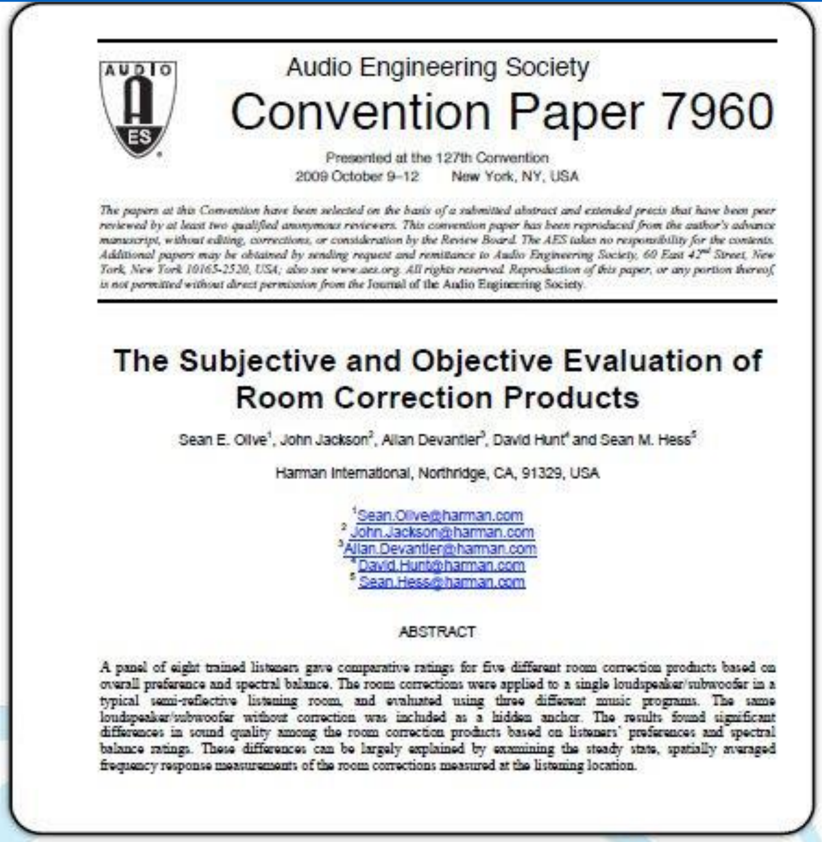
JJ's Conclusions on Room Correction

- At low frequencies, correct the overall room response
- At high frequencies, correct the first arrival timbre
- Always, obviously, correct gain and delay between channels
- Relative correction between channels does more perceptually than the same amount of CPU applied to flattening the system analytically
- Too much correction is bad
- Don't correct for nulls
- Long-window corrections at high frequencies cause the "dentist drill" experience, because the system will be equalized to provide way, way too much correction at high frequencies for the first-arrival signal.

Preferred Room Correction Target Response

http://seanolive.blogspot.com/2009/11/subjective-and-objective-evaluation-of.html





Audio Engineering Society

Convention Paper 7960

Presented at the 127th Convention
2009 October 9–12 New York, NY, USA

The Subjective and Objective Evaluation of Room Correction Products

Sean E. Olive[1], John Jackson[2], Allan Devantier[3], David Hunt[4] and Sean M. Hess[5]

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[4]David.Hunt@harman.com
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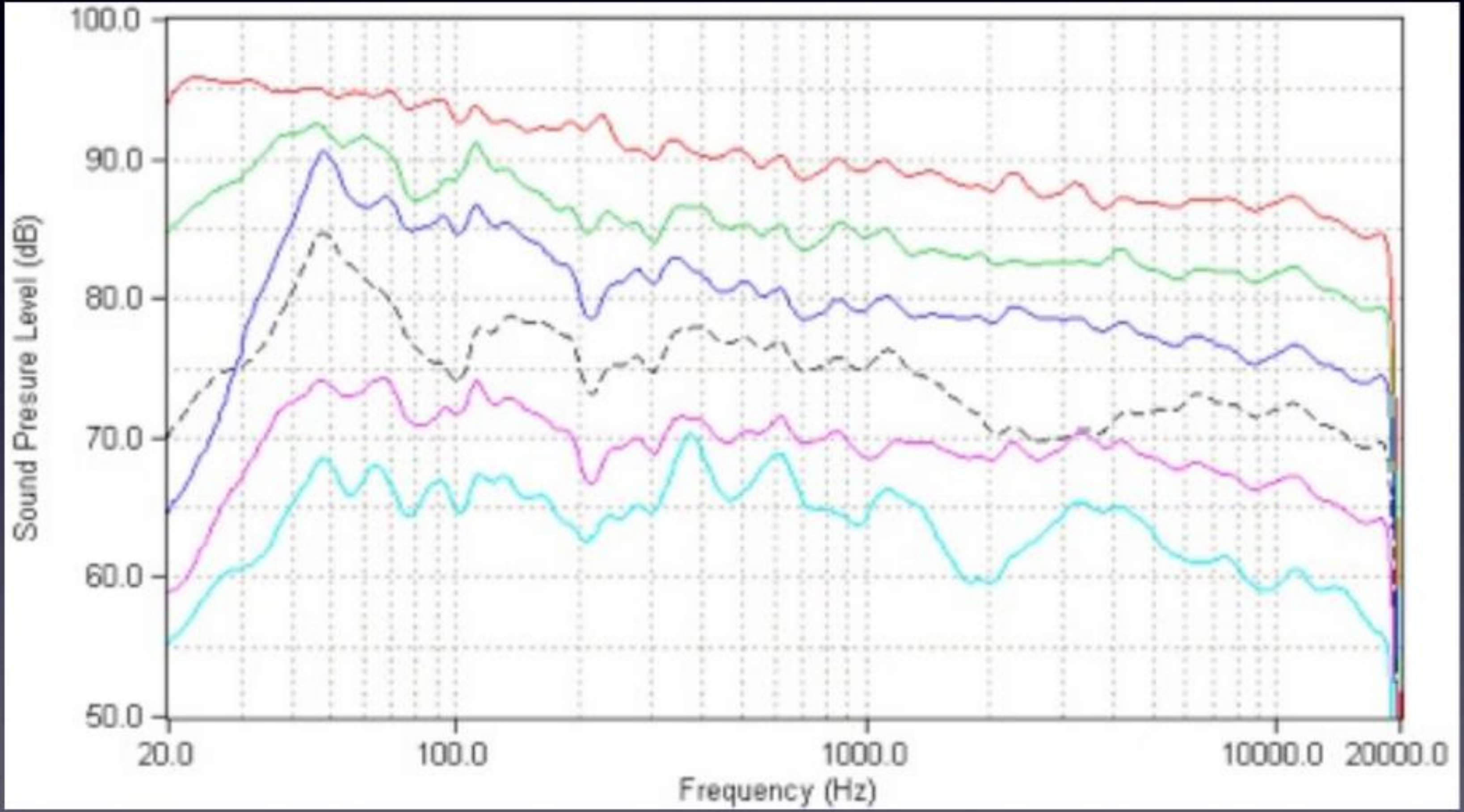
ABSTRACT

A panel of eight trained listeners gave comparative ratings for five different room correction products based on overall preference and spectral balance. The room corrections were applied to a single loudspeaker/subwoofer in a typical semi-reflective listening room, and evaluated using three different music programs. The same loudspeaker/subwoofer without correction was included as a hidden anchor. The results found significant differences in sound quality among the room correction products based on listeners' preferences and spectral balance ratings. These differences can be largely explained by examining the steady state, spatially averaged frequency response measurements of the room corrections measured at the listening location.

Average Magnitude Response at Primary Listening Seat

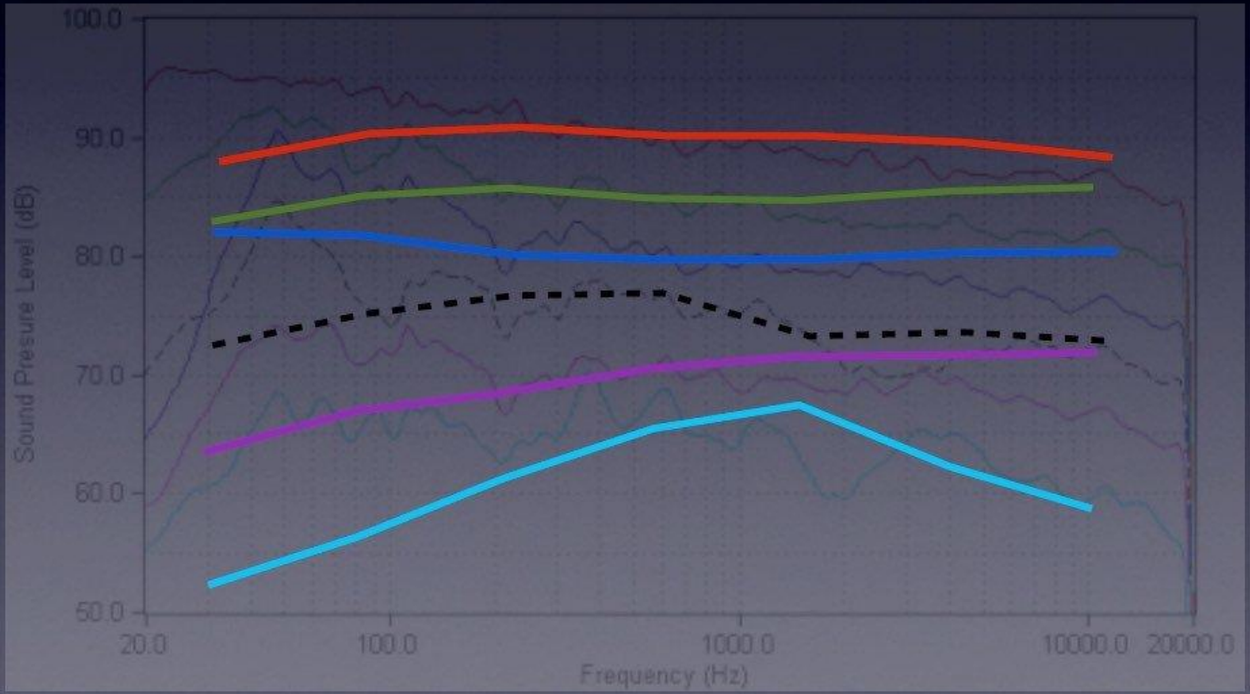
Less Preferred



A Tilted Frequency Response is Perceived as Flat

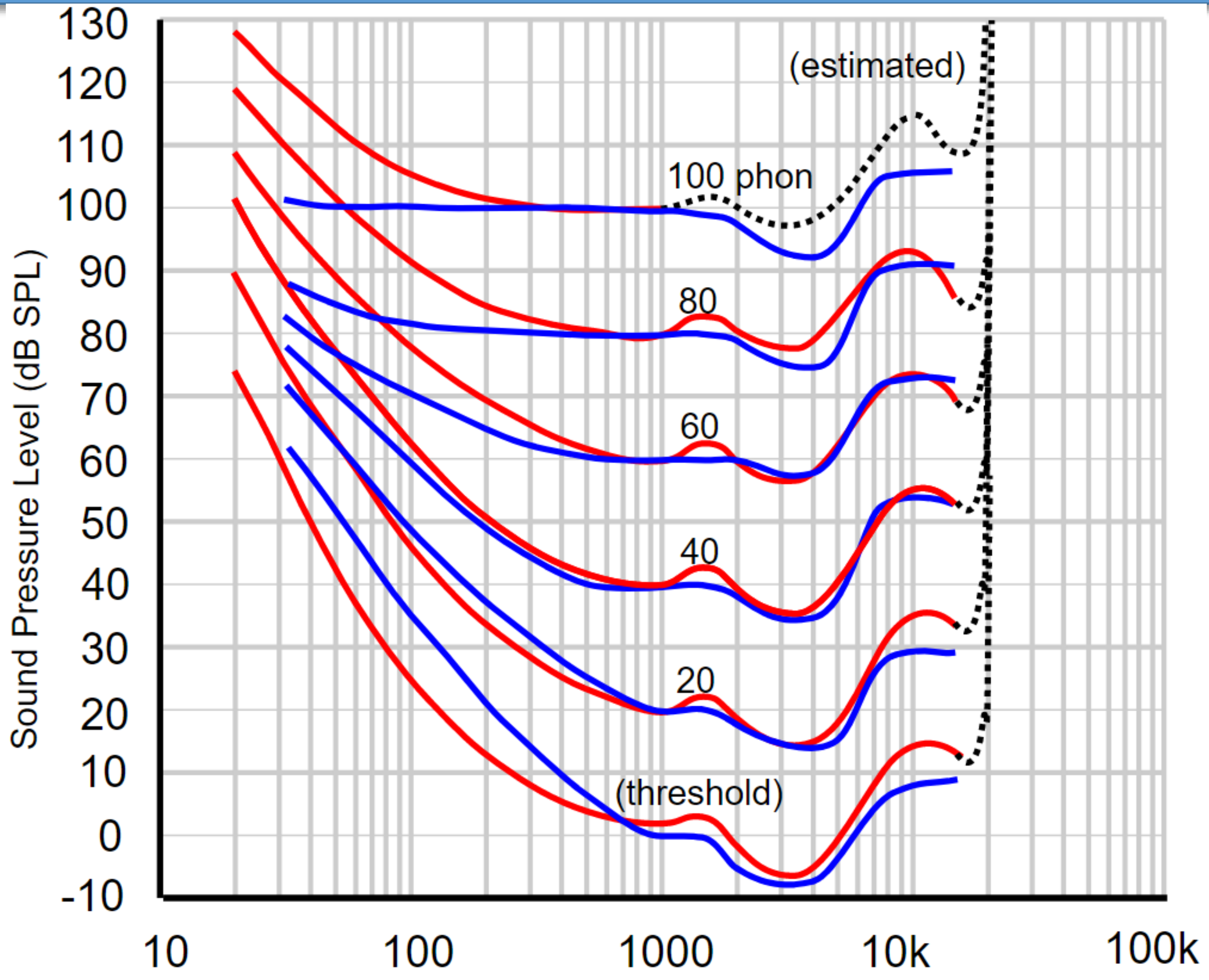
Perceived versus Measured Spectral Balance

Less Preferred



Flat in-room response is not the preferred target

Our perception of FR varies with SPL



Equal-loudness contours (red) (from ISO 226:2003 revision)
Fletcher–Munson curves shown (blue) for comparison

- **<u>At *low* listening volumes</u>** – mid range frequencies sound more prominent, while the low and high frequency ranges seem comparably quieter

- **<u>At *high* listening volumes</u>** – the lows and highs sound more prominent, while the mid range seems comparatively softer

- At ~83 dB SPL C weighting, our ears hear the flattest. 77 to 83 dB SPL is used to mix and master to balance bass with mids and treble

- At lower levels, use a loudness control

The Haas Effect

https://youtu.be/UQOkSF8auFc?t=247

Yes, we can hear a 3ms delay in sound. Mixing engineers often use it as a digital delay trick to "thicken" or "double" a track like a vocal or guitar track.

With some listener training, one can start to hear timing differences. Does require stereo for best effect.

Once you hear it, it can't be unheard ☺

DSP Modelling Using SOTA DRC Software

- Now that we have a good picture of the important parameters, we can start to model/design a solution to restore the ideal loudspeakers minimum phase response to our ears.
- Much like SPICE for electronics, we require FIR filter Designer software that allows us to model these acoustic and psychoacoustic parameters with a high degree of accuracy and precision.
- Currently, there are only a handful of DSP/DRC FIR filter designer software products on the market that can meet all of the modelling requirements.

Steps for DSP Loudspeaker & Room Correction

1. Measure the speakers – single or multiple impulse measurements.
2. Filter the measurement – typically using a psychoacoustic filter and frequency dependent window (FDW). Separate the minimum phase response from excess phase response.
3. Draw a target frequency response. Partial or full range correction.
4. Invert the minimum phase response (see John M REW paper on minimum phase systems).
5. Determine how much excess phase correction is required. Again using FDW and options for partial or full range correction.
6. Generate linear phase FIR filters and package for JRiver, Roon, HLC, etc.
7. Install filters and listen – compare and determine preference.

Acoustic measurements of the filters are within 0.25 dB of the simulations.

FIR Filter Basics

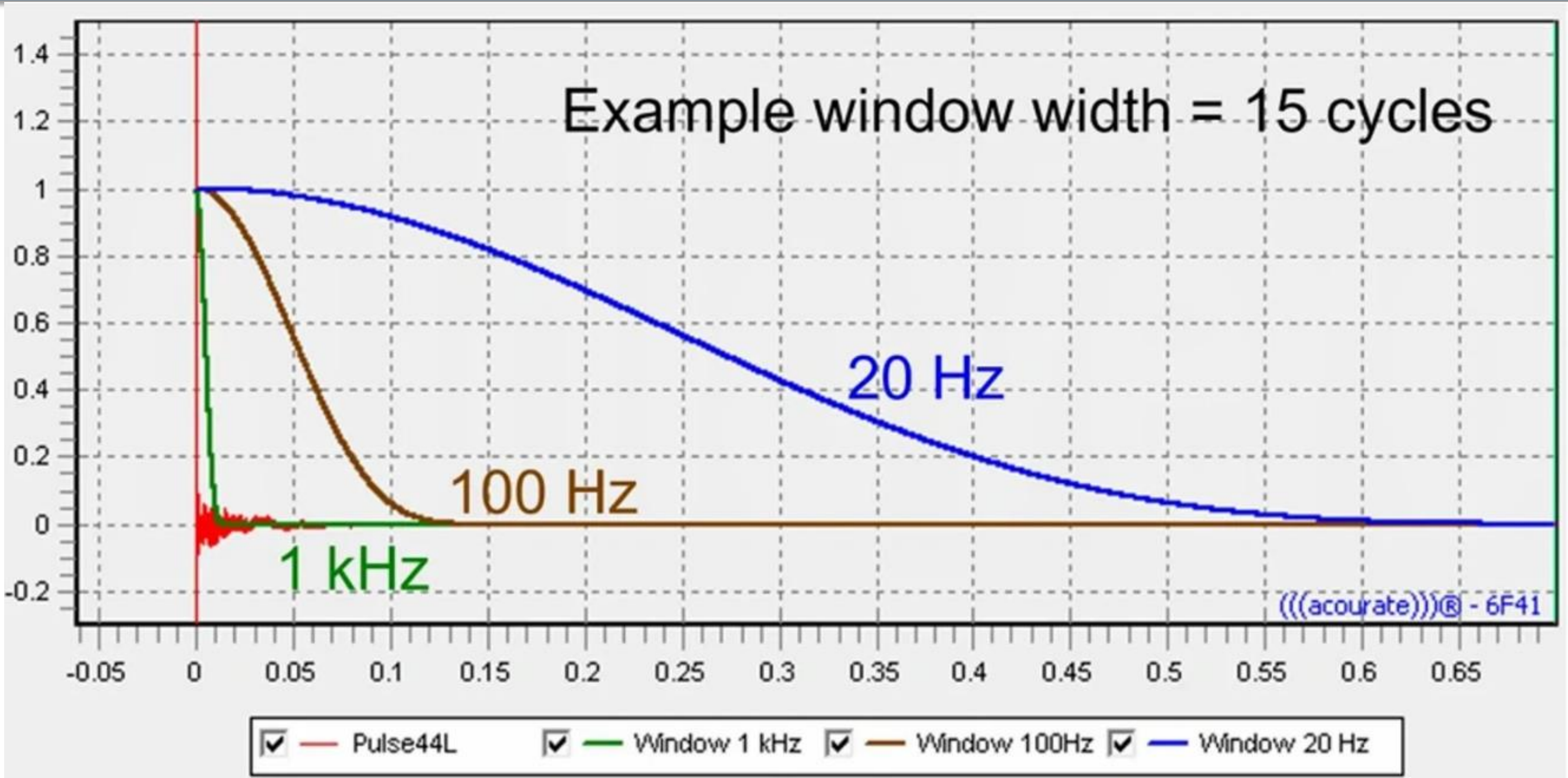
- A FIR filter with 65536 taps @ 48 kHz has a frequency resolution of 48000/65536 = 0.732 Hz.
- The frequency range spans 0 Hz to 24 kHz (fs/2). Thinking of a FIR filter as a graphic equalizer: 24000/0.732 = 32768 sliders for our FIR equalizer.
- IIR filters are minimumphase filters. They cannot adjust the excessphase as an excessphase correction requires a time delay.
- The correction of excessphase is the time reversed excessphase. This introduces a delay which cannot be achieved by IIR filters.
- Linear-phase FIR filters delay the input signal but don't distort its phase.
- The total correction is the convolution of minphase correction and excessphase correction.

Psychoacoustic Filtering

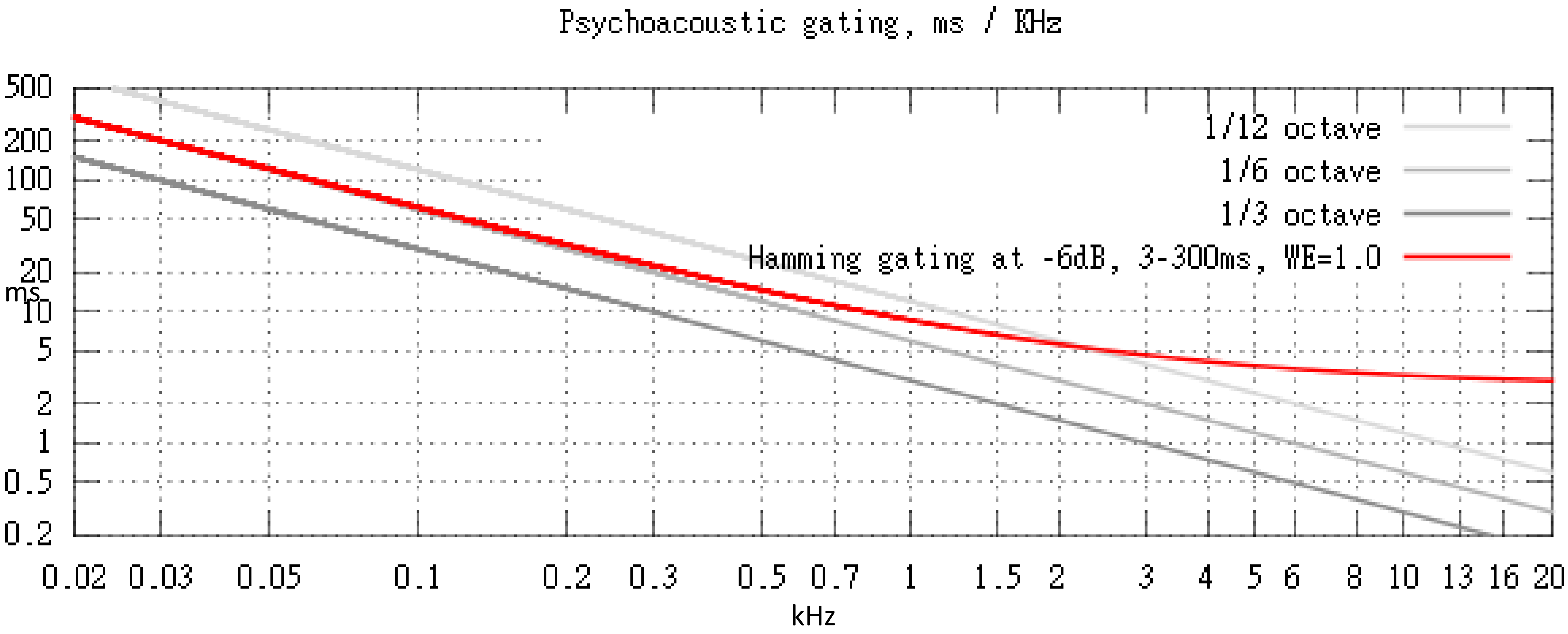
Frequency Dependent Windowing (FDW)

- FDW is a powerful tool for reducing the effect of room reverberations on a measured impulse response.
- The FDW examines the impulse response according to its frequency components, and at every frequency it accepts a window of time that is specified as a number of cycles at that frequency and ignores any portion of the signal at that frequency that occurs outside the time window, thereby ignoring much of the influence of reverberations and reflections that arrive later than the direct signal.
- In SOTA DSP FIR filter design tools, one can specify different lengths of time windows for both low and high frequencies. The values for the time window between the low and high frequencies are adjusted smoothly. Remember that the window is specified in cycles not time.

Frequency Dependent Windowing (FDW)



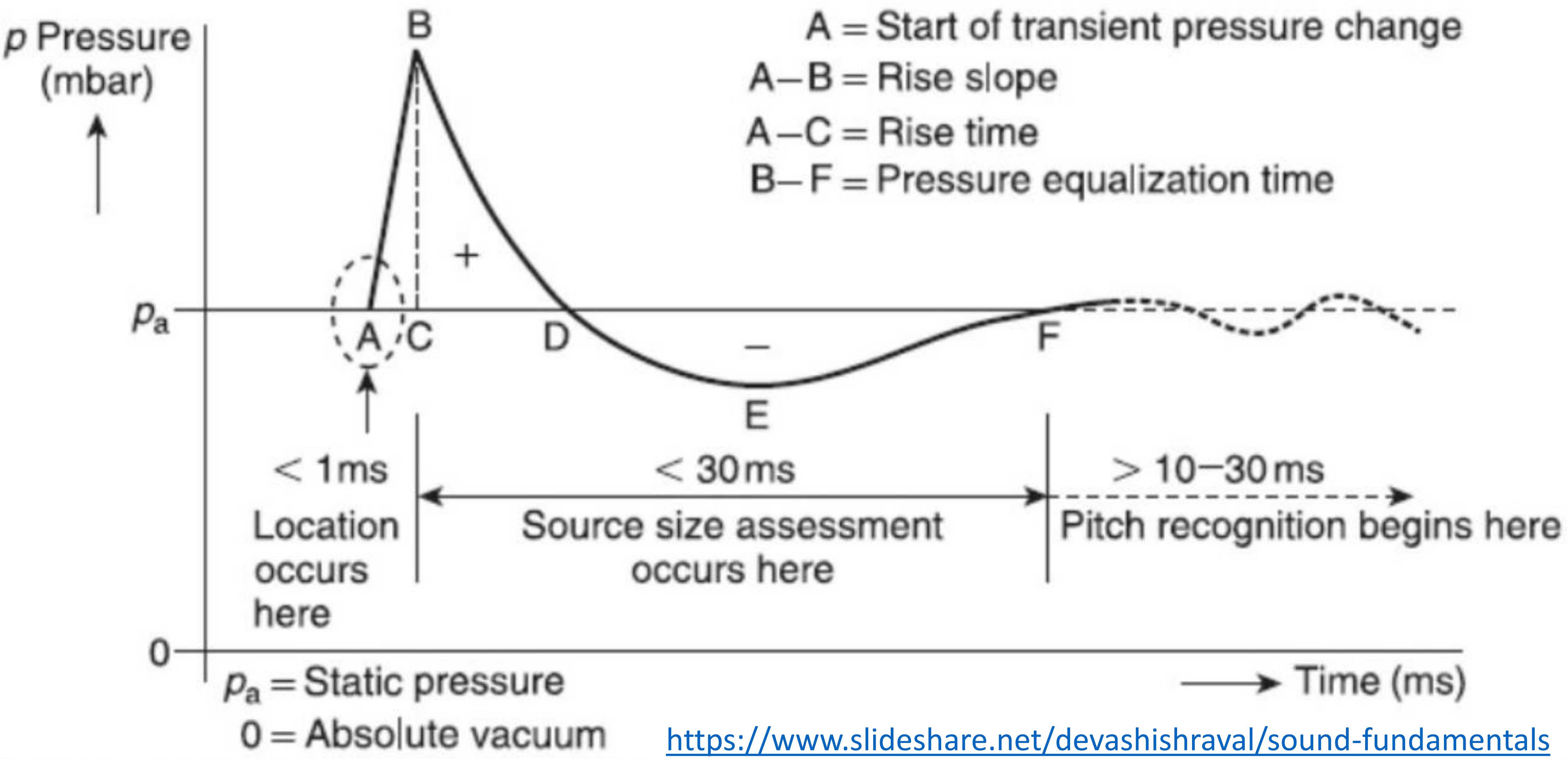
Frequency Dependent Window Example



Calculating FDW Width in Cycles

- As an example of a 15/15 cycles window for both low and high frequencies. Using http://www.sengpielaudio.com/calculator-period.htm
- 20 Hz has a time period of 50ms, so using 15 cycles, a window width of 15 * 50 ms = 750 ms at 20 Hz. At 1 kHz the window width is 15ms and at 10 kHz it is 1.5ms.
- Now using a 6/1 cycles window and solving for a particular frequency like 600 Hz:
- Window width 600Hz = 6 + (1 - 6)/(Log(24000)-Log(4))*(Log(600)-Log(4)) = 2.18 cycles.
- 600 Hz has a time period of 1.6 milliseconds. So at 600 Hz with a 2.18 cycle window we get 2.18 x 1.6ms = ~3.5ms time window. Basically the direct sound of the loudspeaker.
- Therefore at 600 Hz anything outside a 3.5ms window is not included in the correction.
- Sound travels roughly 1 foot per millisecond. So any reflection at 3.5 ft or greater at 600 Hz is not included in the correction filter. As the frequency goes up the time window gets progressively smaller.
- As JJ says: At high frequencies, you're concerned only with the direct signal and the early reflections. This is almost "speaker plus speaker stand" correction.

Transient Response and Pitch Recognition



https://www.slideshare.net/devashishraval/sound-fundamentals

Let's Design a FIR Filter Using These Concepts

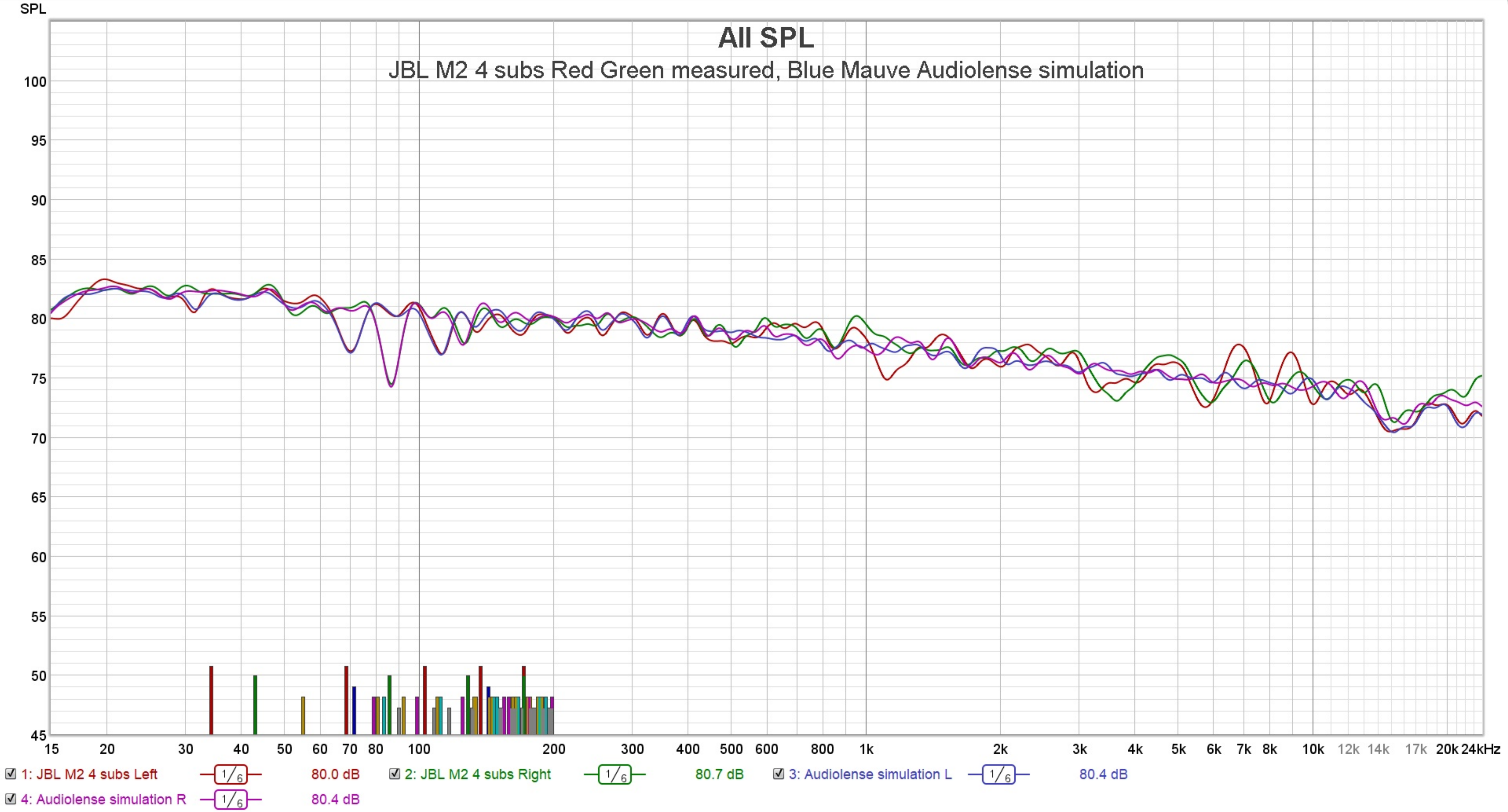
- Walkthrough using a couple of FIR filter designers to create high resolution transfer functions implemented as linear phase FIR filters.

- Install the filters.

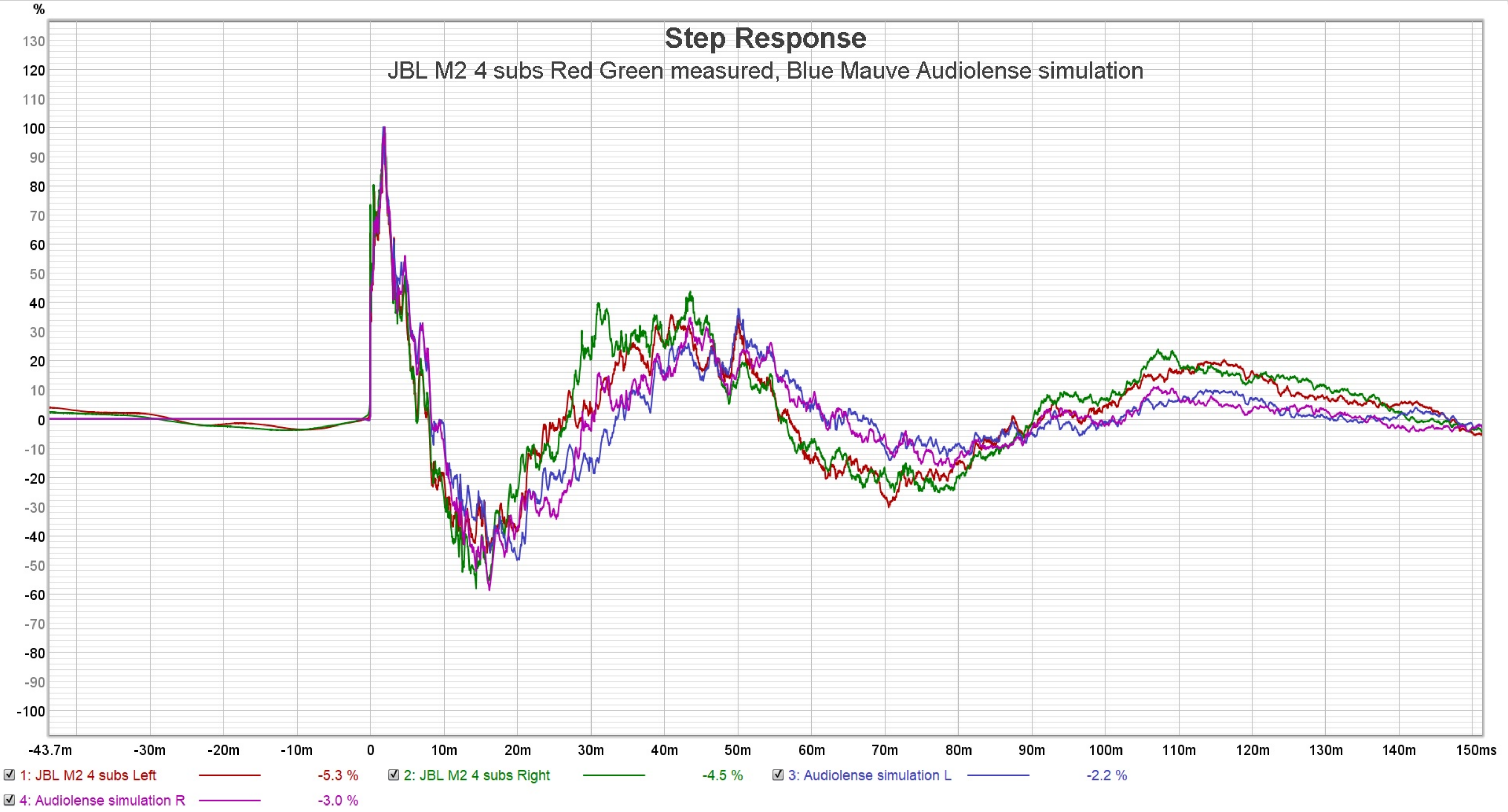
- Comparison listening using HLC.

- Filter design verification: measure the acoustic response with the filter in the system to verify transfer function.

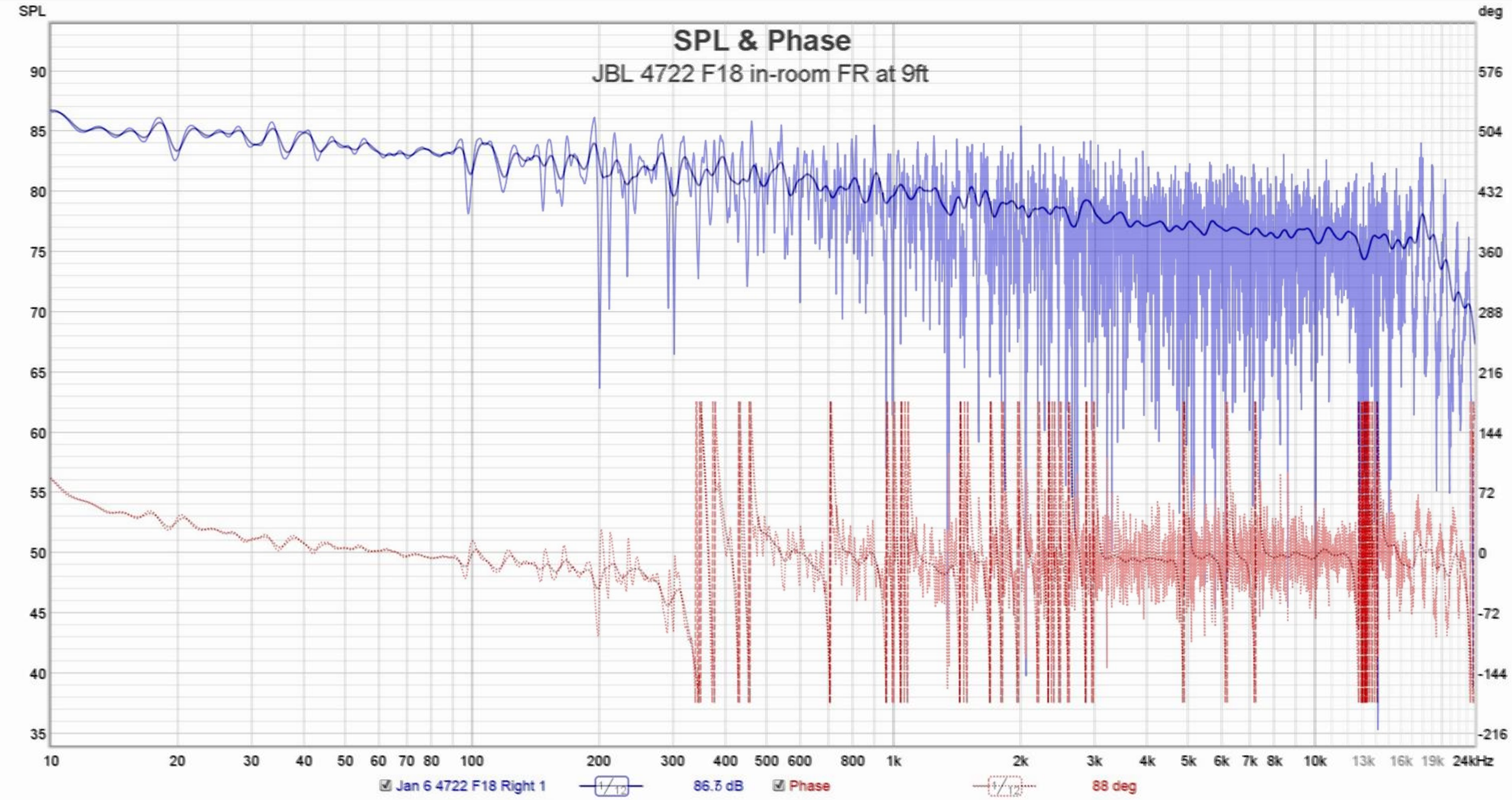
JBL M2 + 4 Subs Transfer Function Verification



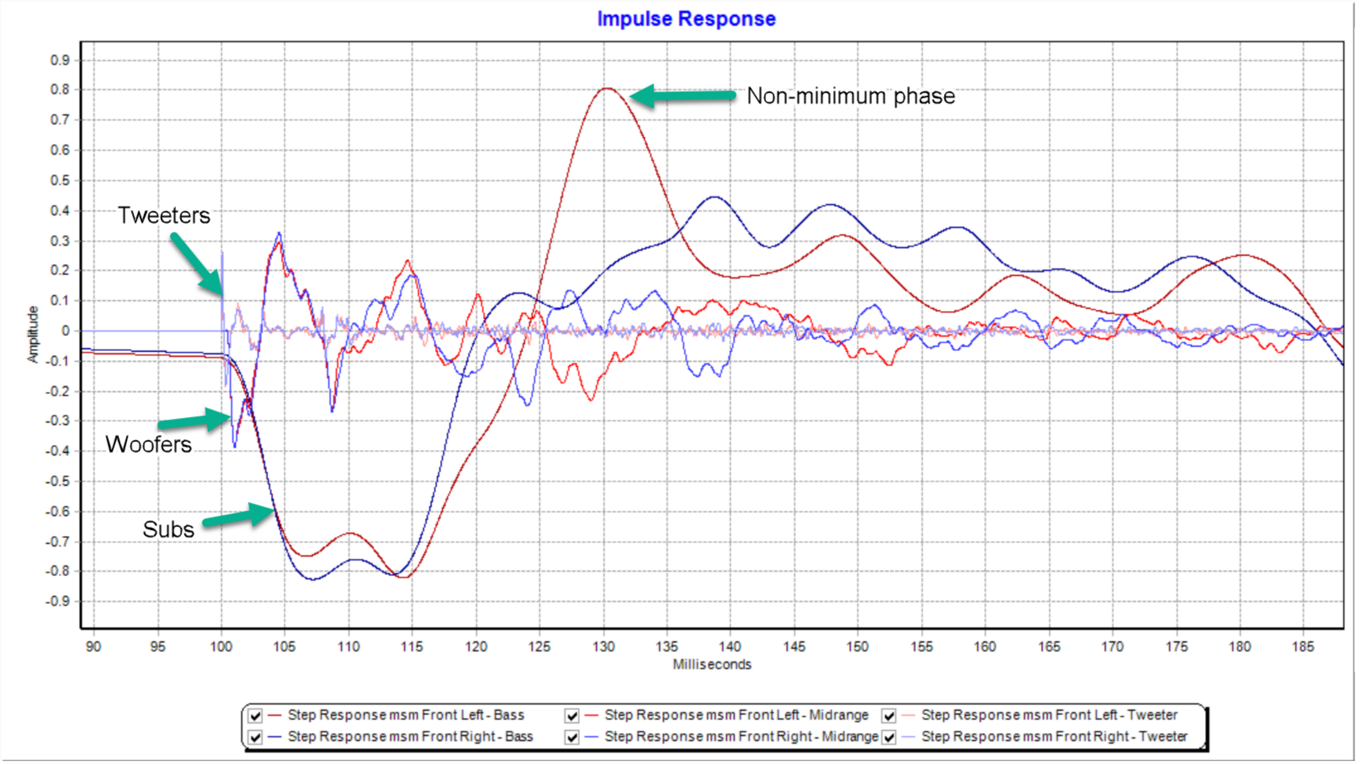
JBL M2 + 4 Subs Transfer Function Verification



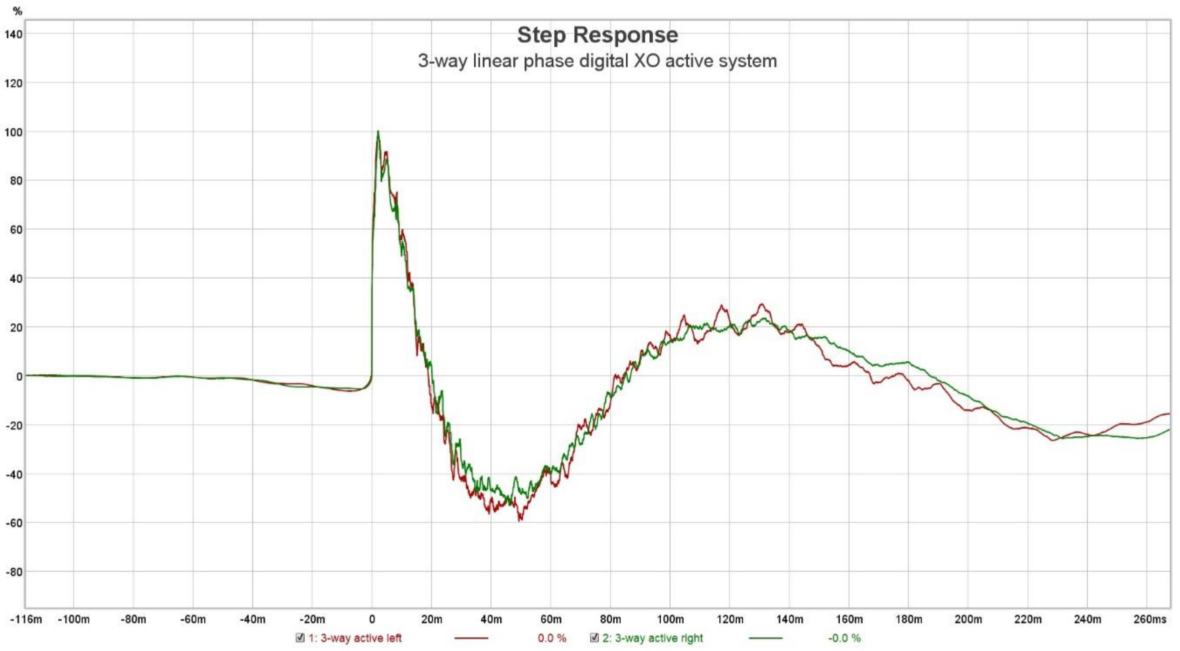
FIR Filter Design verification freq. response



FIR Filter Design Verification Step Response



Before DSP

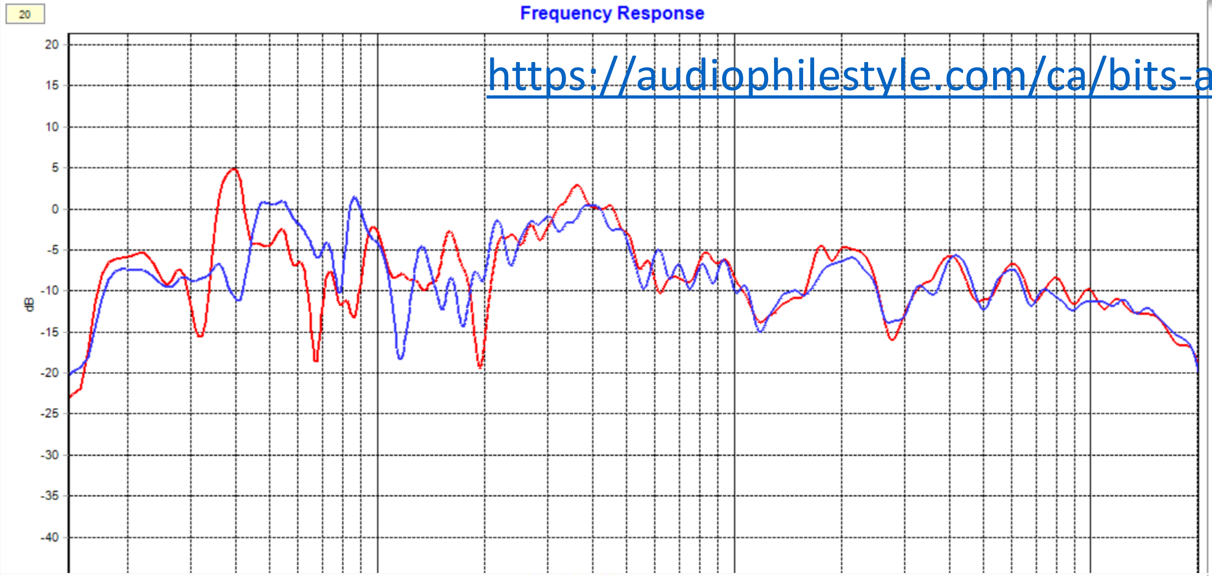


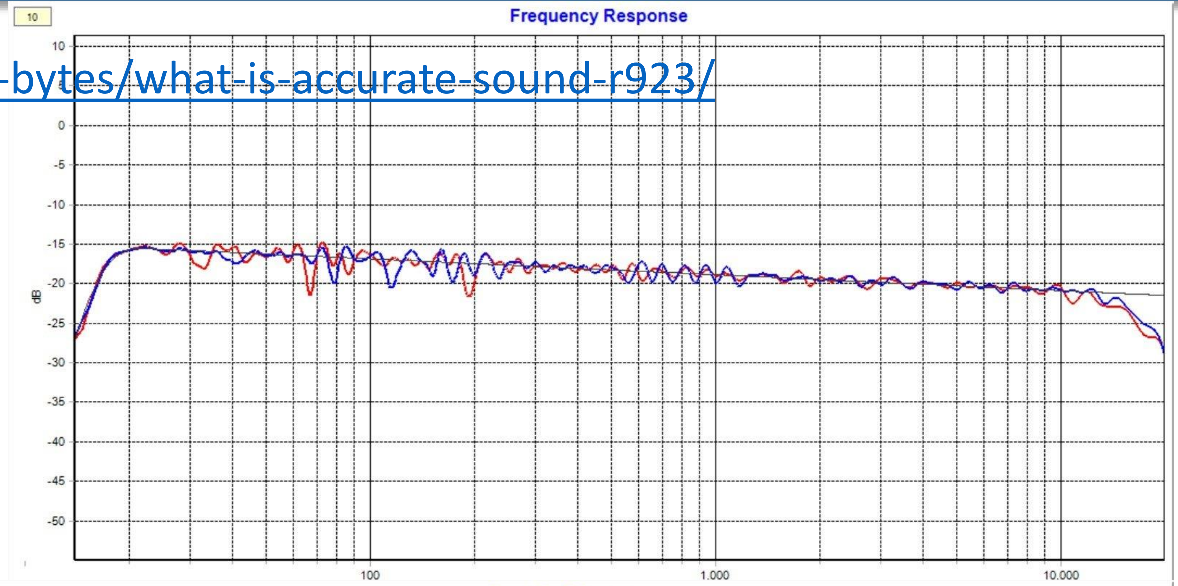
After DSP

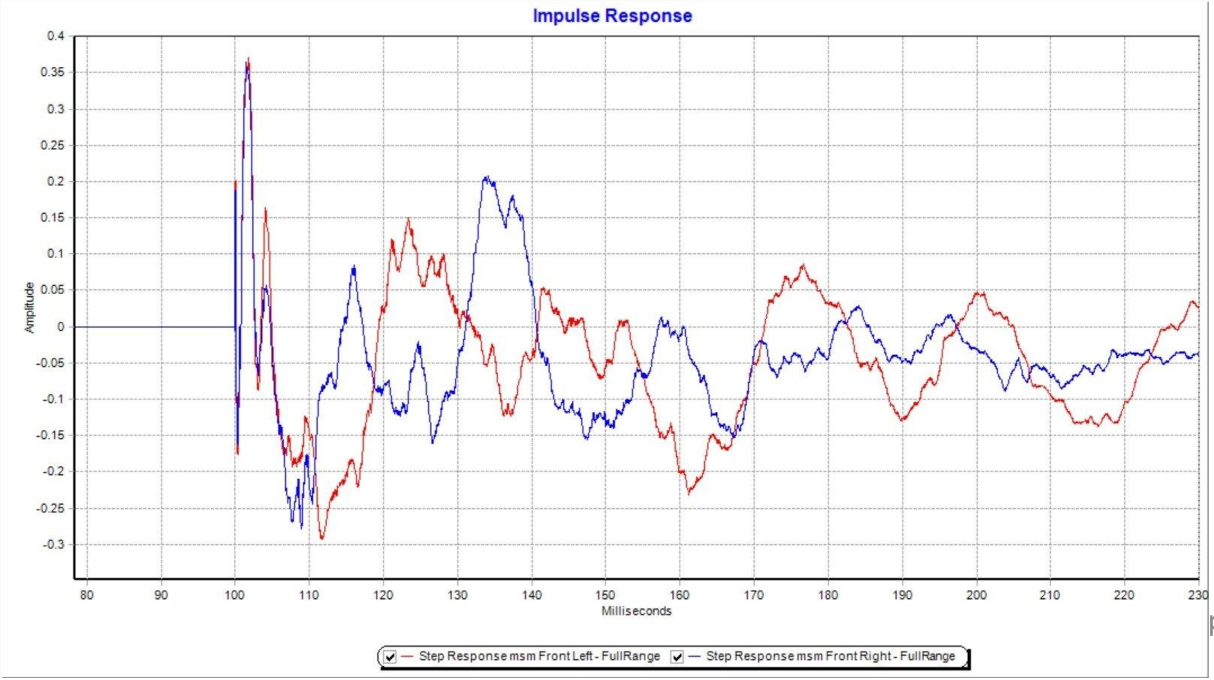
(Same speaker system as previous slide)

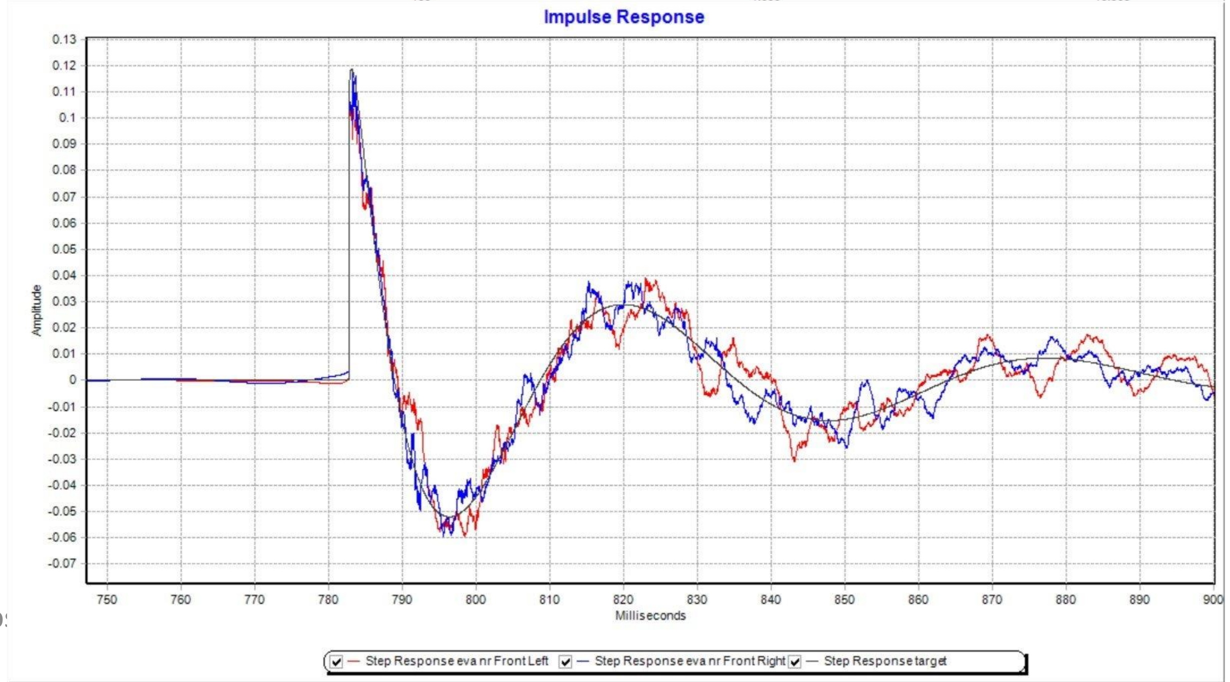
Full Range Loudspeakers Before and After DSP

https://audiophilestyle.com/ca/bits-and-bytes/what-is-accurate-sound-r923/

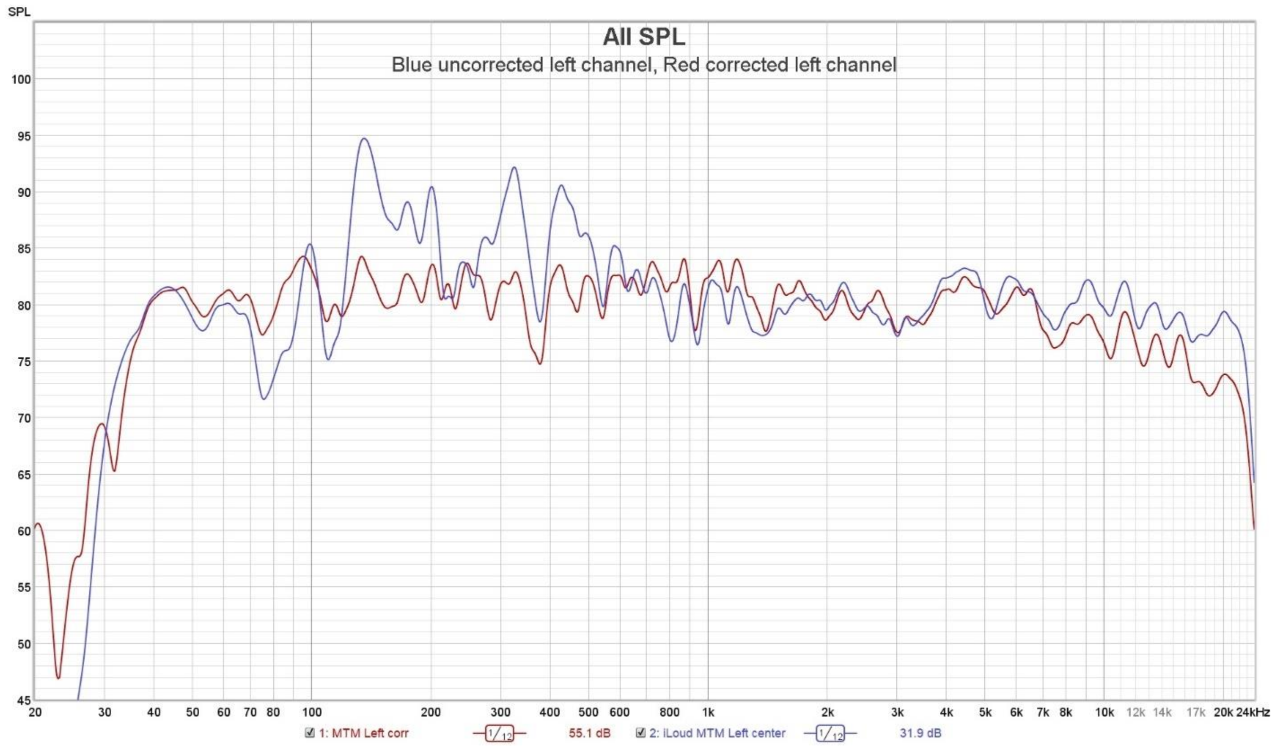






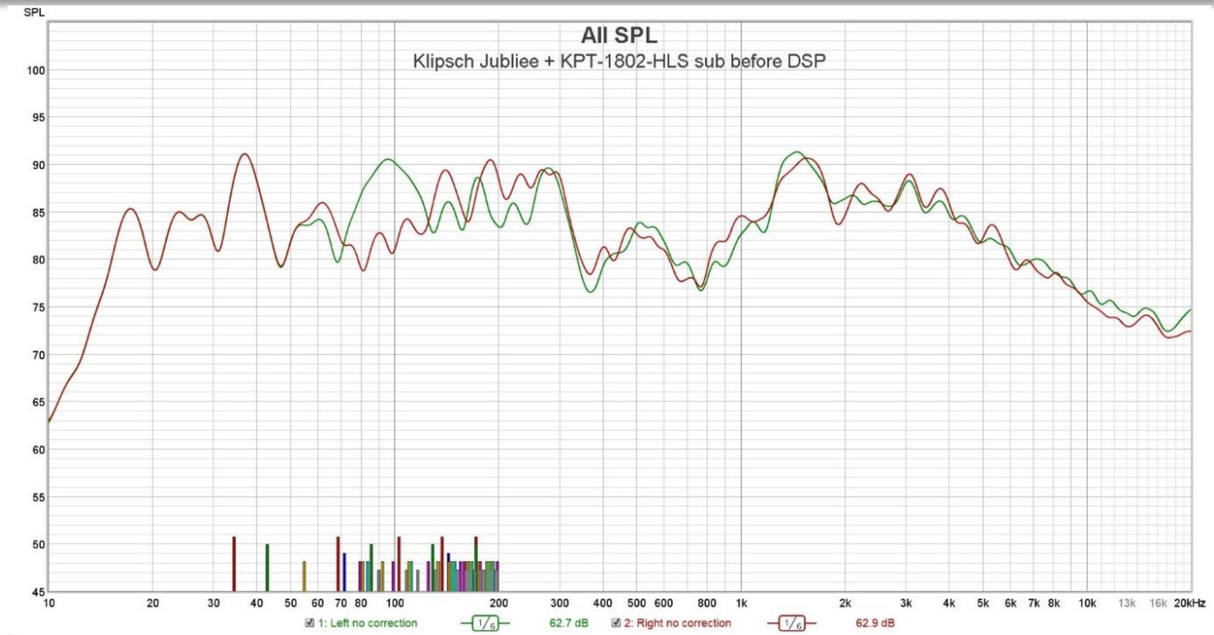


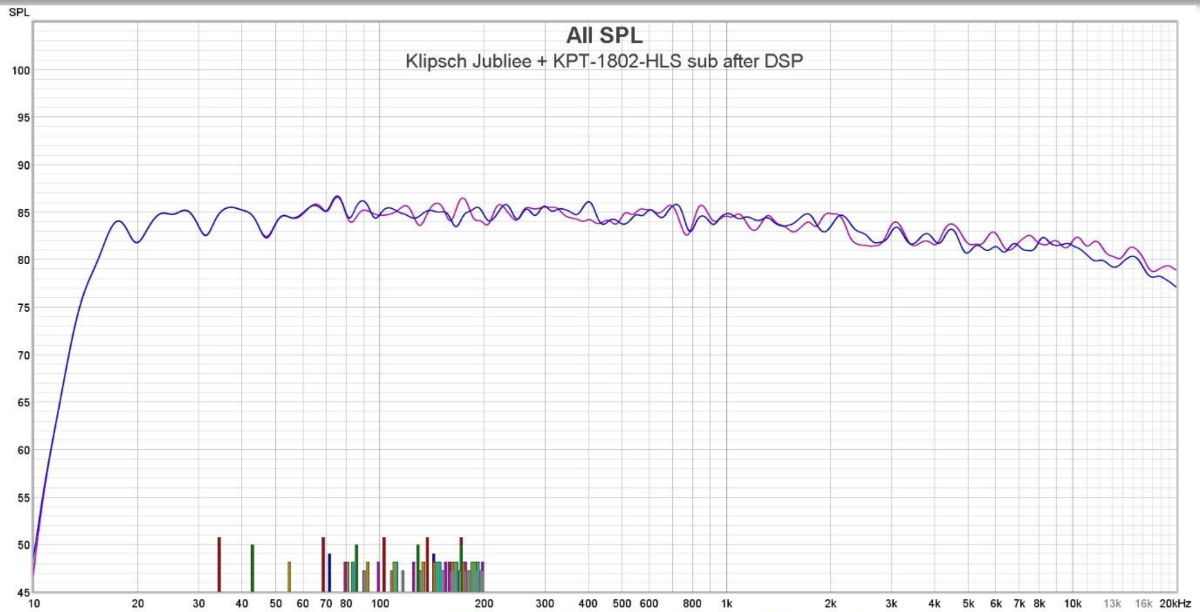
Desktop Speakers Before and After DSP

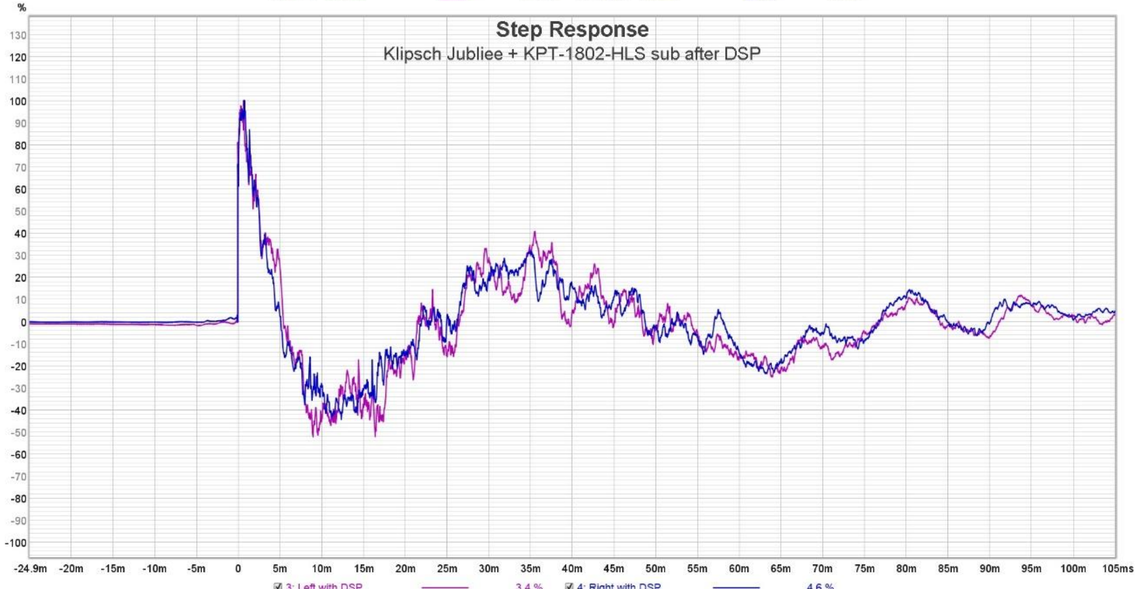


https://audiophilestyle.com/ca/reviews/calibrating-desktop-speakers-using-focus-fidelity-filter-designer-r990/

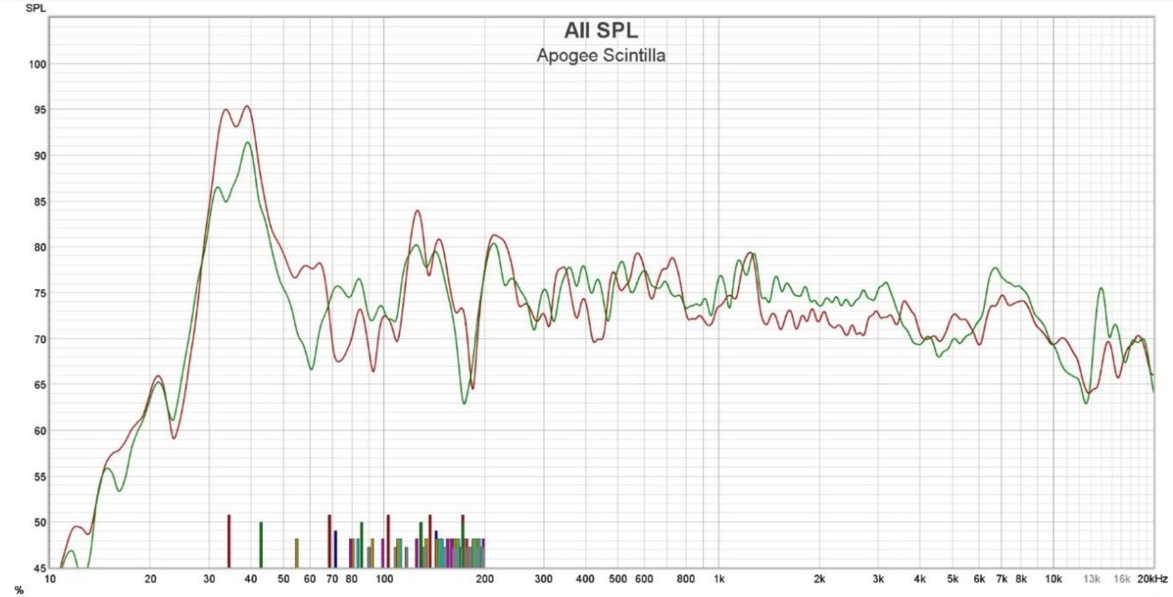
3 Way Digital XO Horn Loaded System Before and After DSP

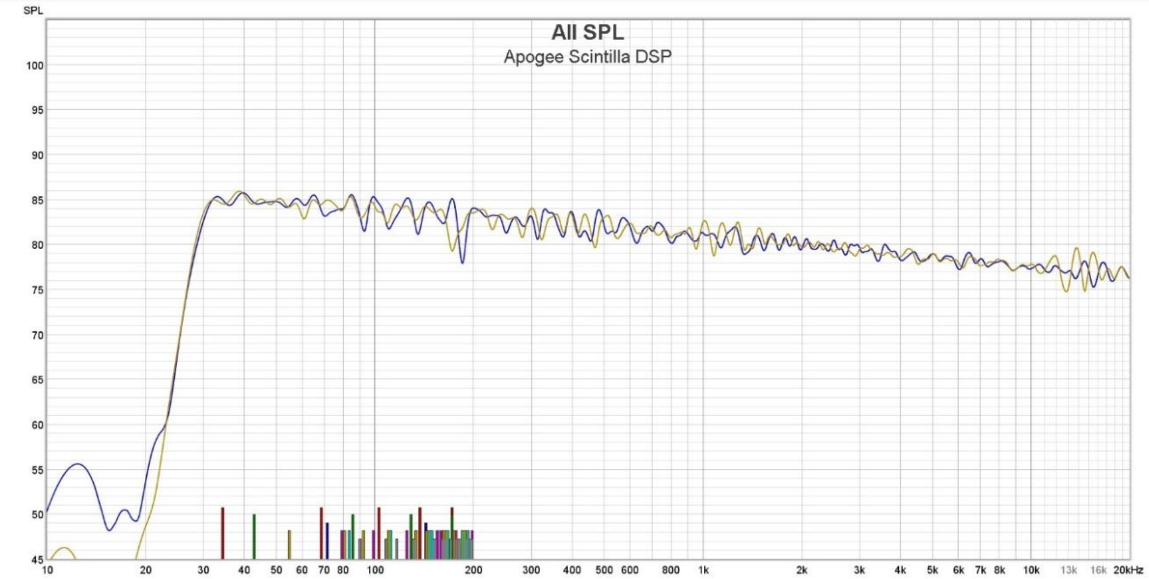


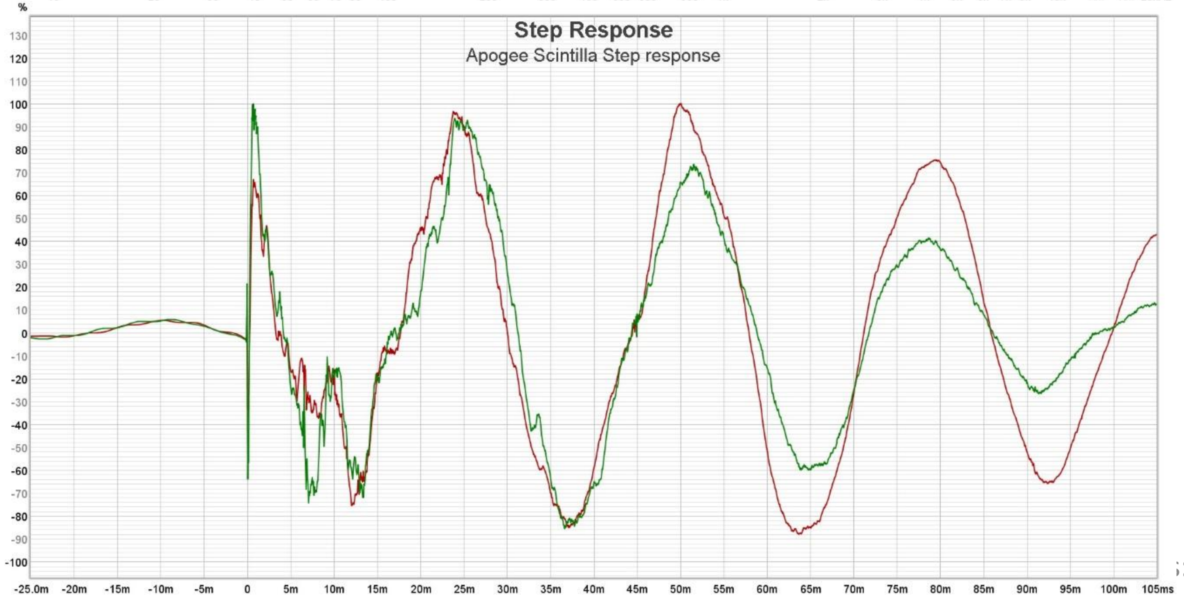


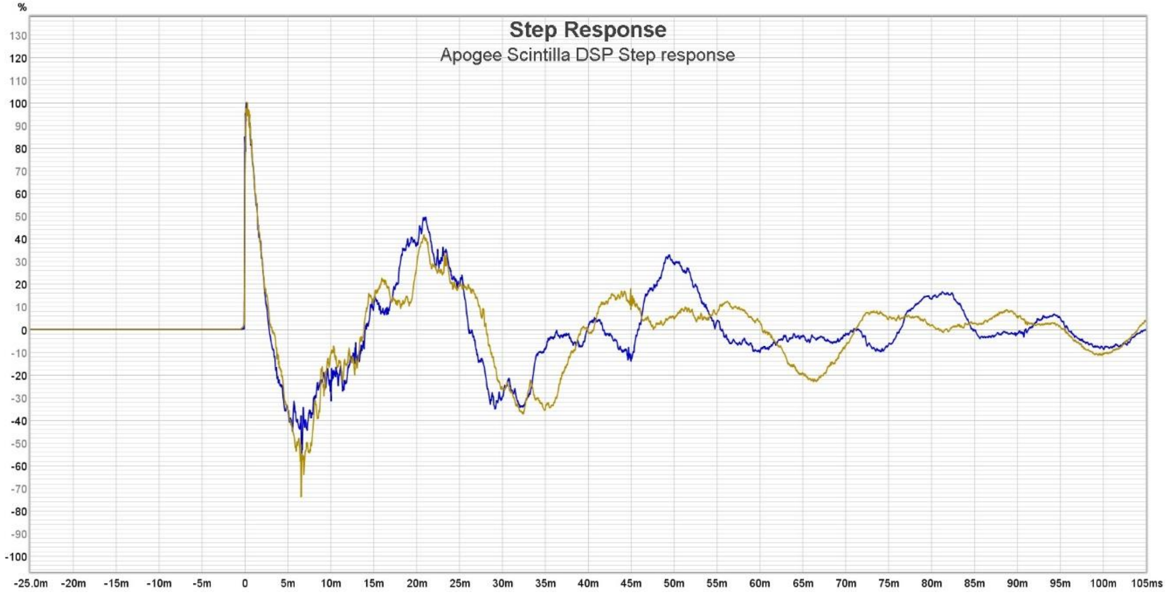


Apogee Scintilla panel before and after DSP

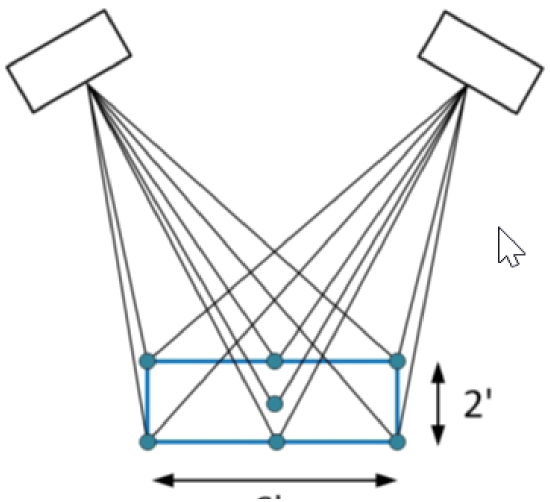




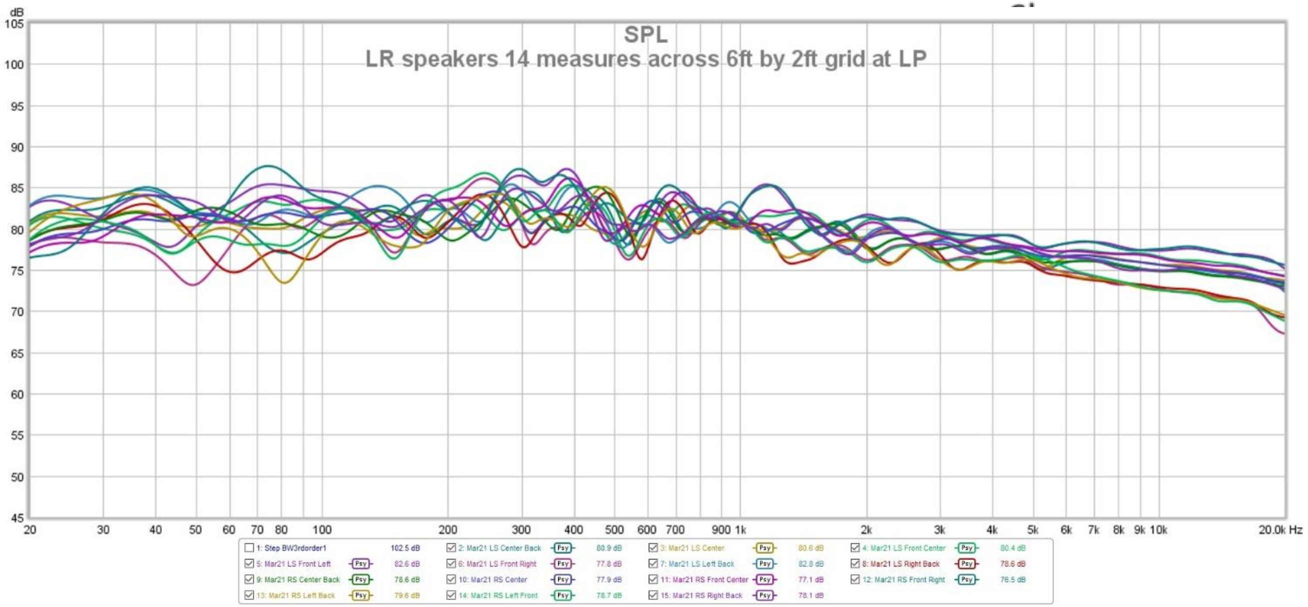


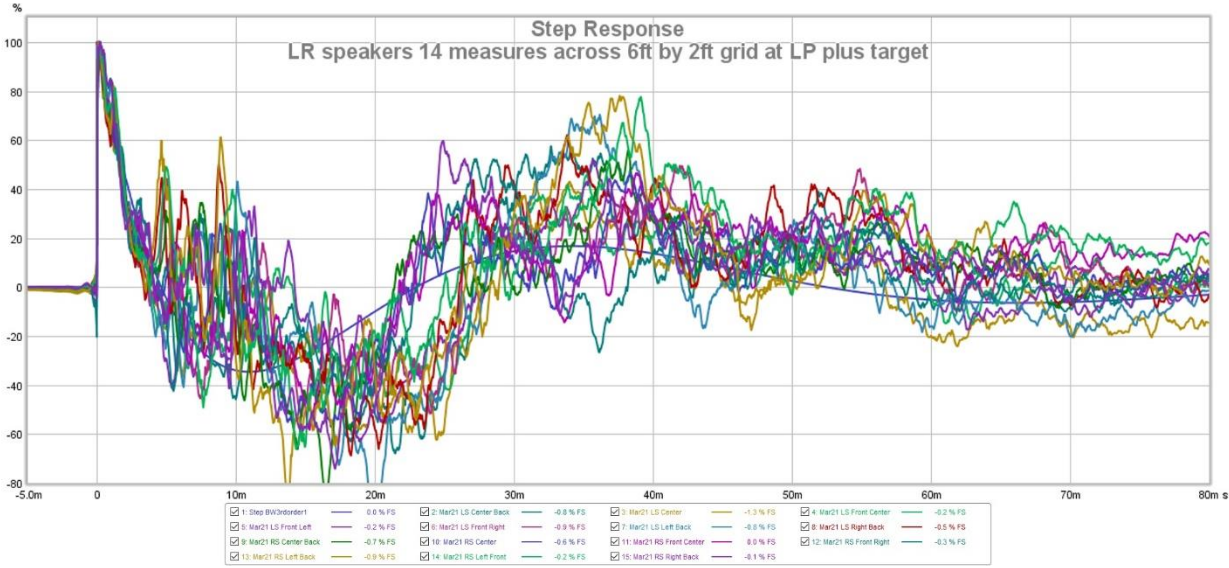


DSP FIR Filter Design Verification – Multiple Locations



Room is 30' x 15.5' x 8'
Listening position offset
to left side of room
Loudspeakers in a 9'
equilateral triangle





https://www.amazon.com/dp/B01FURPS40

Conclusions

- Using the correct DSP model and solution, we can reverse the effects of the room and linearize frequency response distortion in loudspeakers.
- At a minimum, psychoacoustic filtering and frequency dependent windowing is a requirement for good sounding filters.
- The examples and walkthrough demonstrate this using traditional cone and dome speakers, horn loaded speakers, desktop speakers, and panel loudspeakers all in different rooms driven by all different electronics.
- Having DSP'd over 125 different "loudspeakers in rooms" from around the world over the past 18 months illustrates the modelling approach and solution results are robust and repeatable.

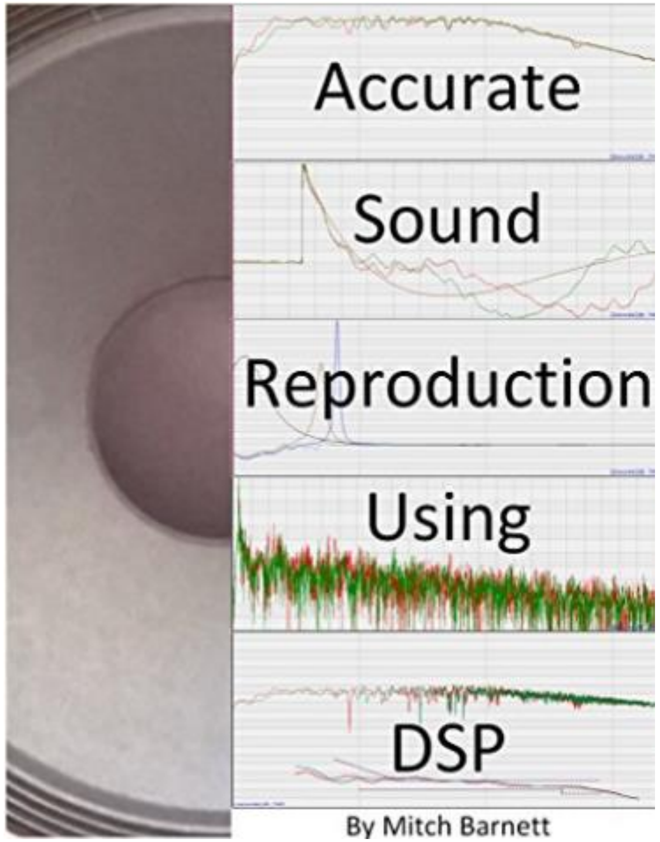
Conclusions Continued

- State of the art DRC/DSP software can be purchased for under US $450.
- Not recommended, a calibrated USB measurement microphone is under US $100.
- Highly recommended, a good analog measurement mic and mic preamp cost US $200 and up.
- Currently, computer software based DSP systems are considerably more powerful than any h/w systems with one or two exceptions.
- Having multiple subs does not automatically guarantee a smoother response. In some cases, the more subs the worse the response, and can be up to 30 dB peak to peak variances depending on where the subs are placed.
- Not all DRC/DSP software is the same. There are only a handful of DSP software products that can meet these modelling requirements.

SOTA DRC/DSP FIR Filter Designer Software

- Acourate
- Audiolense
- Focus Fidelity Designer (multiple measurements)
- DRC Denis Sbragion (open source)

About Me



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