The essentials of the Independent Forum of the HiFi and Audiophiles

Acoustic treatment & Calibration
Rephase - The Tutorial
[pda0] & [Bear]
**(with complements from different [members] of the forums the
contributions of [Vonkarl] for the creation of the original French PDF version and
[Sixto] for this Google translation to English and PDF version)**

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Contents

1 Introduction and genesis

On a pipe of [Maxitonus], and after an effective clearing of [Bear], followed by some complementary experiments of [pda0], it seemed interesting to try to Reorder all information acquired on the free software RePhase.

Since we have neither of them any particular competence in acoustics or digital correction, this tutorial is the result of our real experiences of audiophiles, and some pragmatic deductions for using the software.

Since priority has been given to concrete implementation, the only formal guarantee we give is that there are surely some errors of analysis and / or interpretation in this tutorial, and we thank you in advance for your indulgence.

RePhase is a software developed by POS (its nickname on the various forums), which allows you to generate all kinds of digital filters for different purposes. The literature available on the internet is very important, between forums, blogs and other sites dedicated (MiniDSP in particular), we find all it takes, and more, to know all that can do RePhase, but there is not, to our knowledge of official documentation of this software by its author.

As this forum is a HiFi forum, this tutorial will be limited to showing how to apply RePhase to the classic Hifi with 2 "normal" speakers (with passive filter), in order to use it digital correction of phase and amplitude.

We are talking here about using RePhase as a Room Correction tool, without pretending approach the functionalities of the tool which concern the realization of active filters (Active Cross-Over) which aim to replace the passive filters of the speakers.

Using a MiniDSP processor instead of a PC or Mac is not included here, even if most of the operations to be carried out are likely to be very similar.

The toolkit for starting this work consists of two (free) pieces of software: REW and RePhase, even if they both propose to pay a voluntary contribution (donation) to their author to finance the development.

We will also need a 3rd software allowing the Playing of the music and the use correction pulses that have been created using the 2 previous software.

Several software can do the trick, we will talk here about JRiver, which is available for a small license fee (less than 50 €), and also Roon, whose license is relatively expensive (around € 100 / year, or € 500 for a license of unlimited duration) but the user interface is a model of its kind, and the integration of digital correction is extremely simple and user-friendly. Other playback software is also compatible with digital correction, including Foobar, HQPlayer and probably many others.

The respective role of each of these components is as follows:

- Measure the amplitude / frequency and phase response of a conventional stereo hifi system (REW)
- Correct the acoustic phase of any speaker (RePhase)
- Correct the amplitude / frequency response of the system (RePhase)
- Simulate the corrections made, and thus be able to adjust them, if necessary before implementation (RePhase and REW)
- Optionally, measure the corrections made to verify that they meet expectations (REW)
- Implement the corrections, once created by RePhase, in the form of convolution files embedded in the digital stream (JRiver, Roon, Foobar, etc ...).

In short, do whatever you can do with the software of a Trinnov or Dirac Live, and all this for free (at the cost of JRiver or Roon license, or Foobar close).

And for having tried it on our respective systems, on which we also put in Work a Trinnov Amethyst and Dirac, it is clear that the result is quite amazing!

Of course, as our friends across the Atlantic say, " *there is no such thing as a free lunch* ", and therefore the counterpart of the free is a relative complexity of implementation that this tutorial will try to reduce as much as possible. More specifically, let's say that it is the functional richness of the tool, and the author's bias to allow access to the set of correction parameters, which impose this relative complexity.

It should be noted, however, that while it is not necessary to have the slightest acoustics competence to use these software, it is strongly recommended to be in comfortable with a PC or Mac and installing software.

Even if there is no technical difficulty in implementing this software and if the operations to be carried out are quite simple, obtaining a convincing result will require a few hours of effort and patience. But, the brave ones who will go through with this installation effort will be rewarded and will probably have trouble going back! Just like those who, before them, have tasted joys of Trinnov (or Dirac) and cannot really do without it!

What "audiophile" benefits can we expect from RePhase?
As its name suggests, RePhase has been written with the ambition to allow audiophiles to correct the acoustic phase of their speakers.

In the case that we limit ourselves here, the speakers are taken as an inseparable whole. The correction will therefore be applied in the form of a pulse (it is a simple file in .wav format) which will be 'added' to the musical signal by an mathematically complex operation (called convolution) which, fortunately, it is not necessary to understand the detail to use it!

The audible result of this correction will be to ensure a better temporal coherence of the speaker, that is to say that the time differences introduced in the reproduction of the different components of the music by the construction of the speakers will be corrected, so as to that the highs reach your ears at the same time as the corresponding bass. A sound composed of several frequencies will be more coherent. The attacks will be sharper, the separation between instruments and voices will be better and the sound stage will be better reproduced, more coherent. It will gain in depth and realism. For those interested, more info on this topic of phasing on the following thread: Temporal: phase / frequency response

But RePhase does not stop there and allows at the same time, to make corrections at the different frequencies (digital equalizer) and will therefore serve in a certain measure to compensate the colorations of your system. In any case, the effect produced by RePhase is very sensitive to listening and is also perfectly measurable.

Everyone can decide how useful it is on their own system, but we think they can be interesting to try it before any search for optimization via esoteric cables or not. Which obviously does not exclude cable optimization, of course.

As you will realize, beyond the relative complexity of implementation, which goes with the great flexibility provided by this software, there are also several ways to use it according to its sensitivity of audiophile (and / or technophile).

For those who would like a more "general" view of digital correction, the thread next gives a first light and places RePhase in the middle of alternative solutions available on the market: Digital correction

And finally, to situate the use of digital correction in the development of a system, it can also be useful to take a look at this thread: 7 steps to install a system with digital correction.

2 Configuration required

Because RePhase is only available on Windows, a Windows PC is essential for the creation of correction pulses.
But the pulses created by RePhase can then be used on any platform (Mac, Windows, Linux) since it only depends on the software used (JRiver, Foobar, Roon, ...).

You can download RePhase here: https://sourceforge.net/projects/rephase/ The RePhase thread on this forum is here: RePhase

REW is available on Windows and Mac OSX. Both versions work well and can be used indifferently for measurement before and after correction and for all simulations of corrections.

You can download REW here: http://www.roomeqwizard.com
The REW tutorial on this forum is here: REW Tutorial

For those who do not want to use REW, you can also use HOLMImpulse (Windows only) that has a more "rustic" interface and fewer possibilities display, but which includes all the functions necessary for the measurement and simulation, so everything we need here.

You can download HOLMImpulse here: http://www.holmacoustics.com/holmimpulse.php
JRiver is available on Linux, Windows or Mac OSX.

You are free to choose the version that suits you. This is essentially about dealing with digital files. So we will need a computer that will have access, in a way or another, to the audio files to reproduce and on which you will have installed JRiver as player for the future.

On fire blue, [ThierryNK] had opened a post "Why I went to JRiver?" Which is available in the archives and explains all the merits in detail: https://web.archive.org/web/201503180046...37265.html

You can download JRiver here: https://www.jriver.com/download.html JRiver is usable for free for 30 days, so you can try it out, like offered here with RePhase, free of charge. The purchase price is 50 US $ (a little less than 45 €) for a platform or 70 US $ if you want to use it on multiple platforms (Mac, Windows, Linux).

JRiver is particularly practical for several reasons (quality convolution engine on 64bits, and possibility to work as a server and control a DLNA mode reader in particular) but nothing prevents to use another software player, provided that it supports a convolution plugin.

As already mentioned, Foobar, HQPlayer or more recently Roon integrate a motor of convolution natively accepting RePhase pulses. Roon, despite its high price, deserves attention because of its very successful interface, and its convolution engine of high quality and remarkable simplicity of use.

In the end, it is perfectly possible to measure on a Mac, to produce the corrections on Windows (possibly on this same Mac in Windows emulation (via a virtual machine)) and install the fixes on a NAS under Linux. Nothing opposes it technically.

As you understand, the **entire process** will consist of: 1. Take phase and amplitude measurements using REW (or HOLMImpulse) 2. Use these phase and amplitude measurements to correct phase and / or amplitude (totally independently of each other) and create a correction pulse with RePhase. 3. Use this correction pulse in JRiver (or Roon, or Foobar, or other) in a so-called convolution process, to effectively correct the phase (time alignment of the speakers) and / or amplitude.

It should be noted that while this process may seem a bit complex to implement, effective use of the selected correction is then completely transparent and can be completely forget about it. The only constraint is to use JRiver or Roon or Foobar as reader of his dematerialized musical base.

But JRiver or Roon being one of the best players available on the market, it's a constraint that can surely be accommodated.

3 Phase and amplitude / frequency measurement

This is the first step and surely the most delicate! Indeed, this operation is absolutely essential and must be carried out with the greatest care. It is from these measurements that the corrections will be calculated and therefore any error at this point can only lead to a random result and very likely disappointing!

We strongly advise the careful reading of the REW tutorial (reference given above), as well as the page specific to the measurement of the phase which is here: Customized tutorial of the Phase.

3.1 Initial measurement and placement of the microphone:
As we are in the context of digital correction, it is necessary to place the microphone absolutely equidistant from the 2 speakers, sweet spot and ear level.

The use of a laser meter is very practical, but any craft method (with a wire attached to the microphone we walk from one speaker to another for example, or a simple tape measure) will work too.

[pda0] prefers to place the microphone upright (microphone pointing upwards), but [Bear] uses a horizontal position oriented towards the speakers. The important thing is to use the file calibration of the corresponding microphone in REW (see tutorial).

It is essential to have a reference time, and therefore do not forget to activate the appropriate function in the menu of REW Preferences-> Analysis-> Use Acoustic Time Reference.
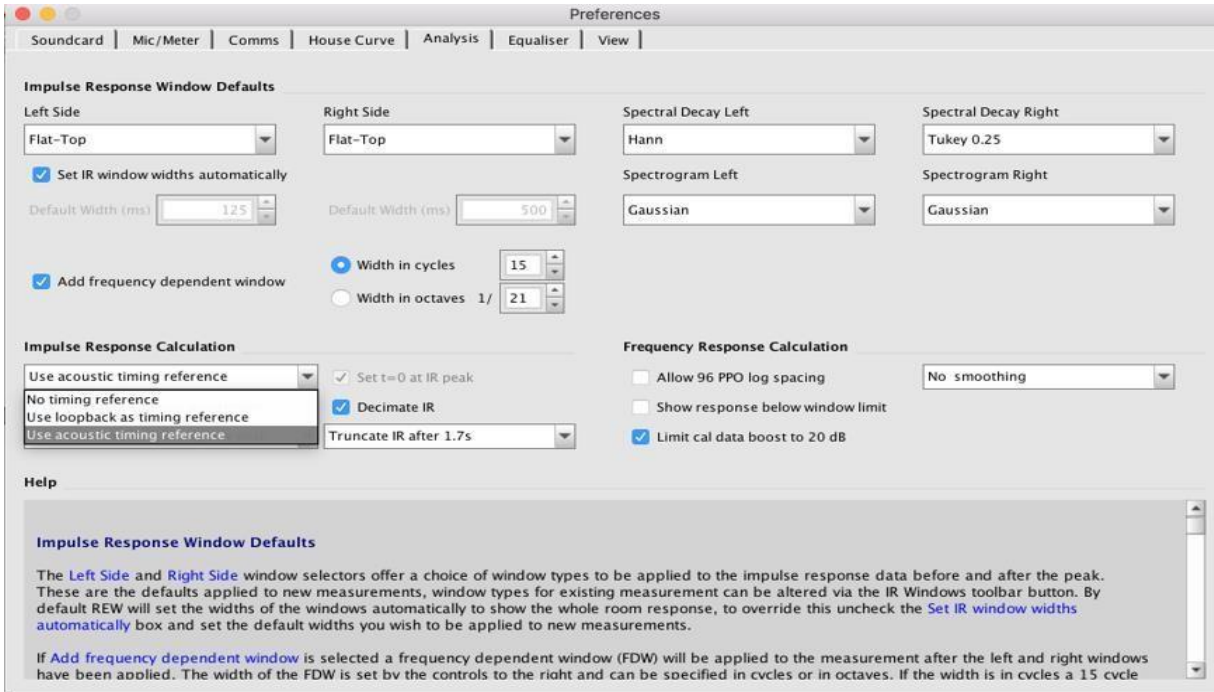
Using this option, REW will emit a sync signal that the microphone is going to save and that will most often automatically stall the measurement window in time. *(It will, despite everything, often be necessary to make a small editing manual zero time, but the acoustic reference is already 95% of the work).*

Once everything in place, we must begin by making 3 successive measurements:
- the left speaker alone,
- the right speaker alone,
- and finally, the 2 speakers together.

(See REW tutorial if applicable).

Note: it is important (because it facilitates the automatic setting) that the acoustic signal of reference either on the left channel for the left speaker and on the right channel for the speaker right (this is manually selected at the time of measurement in the appropriate menu on the measurement window). For measuring the two channels together, the signal to the right or to left does not matter, so we can take any one.

3.2 Timing of the impulse

Once the 3 measurements have been made, we will look at each speaker successively. The measuring the 2 speakers together only to identify the behavior in situation "Normal" response in the bass (the 2 speakers can cause interference between them and create nodes and bellies that do not appear on each speaker taken separately).

The set of procedures below must therefore be carried out twice (once for each speaker).
The first step will consist in precisely stalling the zero time of the pulse for each speaker.

The REW tutorial on phase measurement explains how to proceed and specifies that stall on the first positive peak of the impulse which is that of the tweeter. To reread if necessary: Tutorial Phase Measurement

In case of doubt on the right peak to remember (impulses are a little complex), it is necessary to visualize the pulse in the form of STEP in REW (by ticking the appropriate box on the display of pulse) and hold the peak closest to the first vertical bar (or high because it is not necessarily always very vertical).

Note: If the STEP goes down (instead of upwards as it should), it means that the phase is reversed (it happens because some electronic invert systematically – YBA for example), in this case, it is convenient to invert the pulse to the display in order to find it easier (for that click on the parameters at the top right and check the case 'invert impulse').

See examples below: Apertura Thema (impulse and phase)
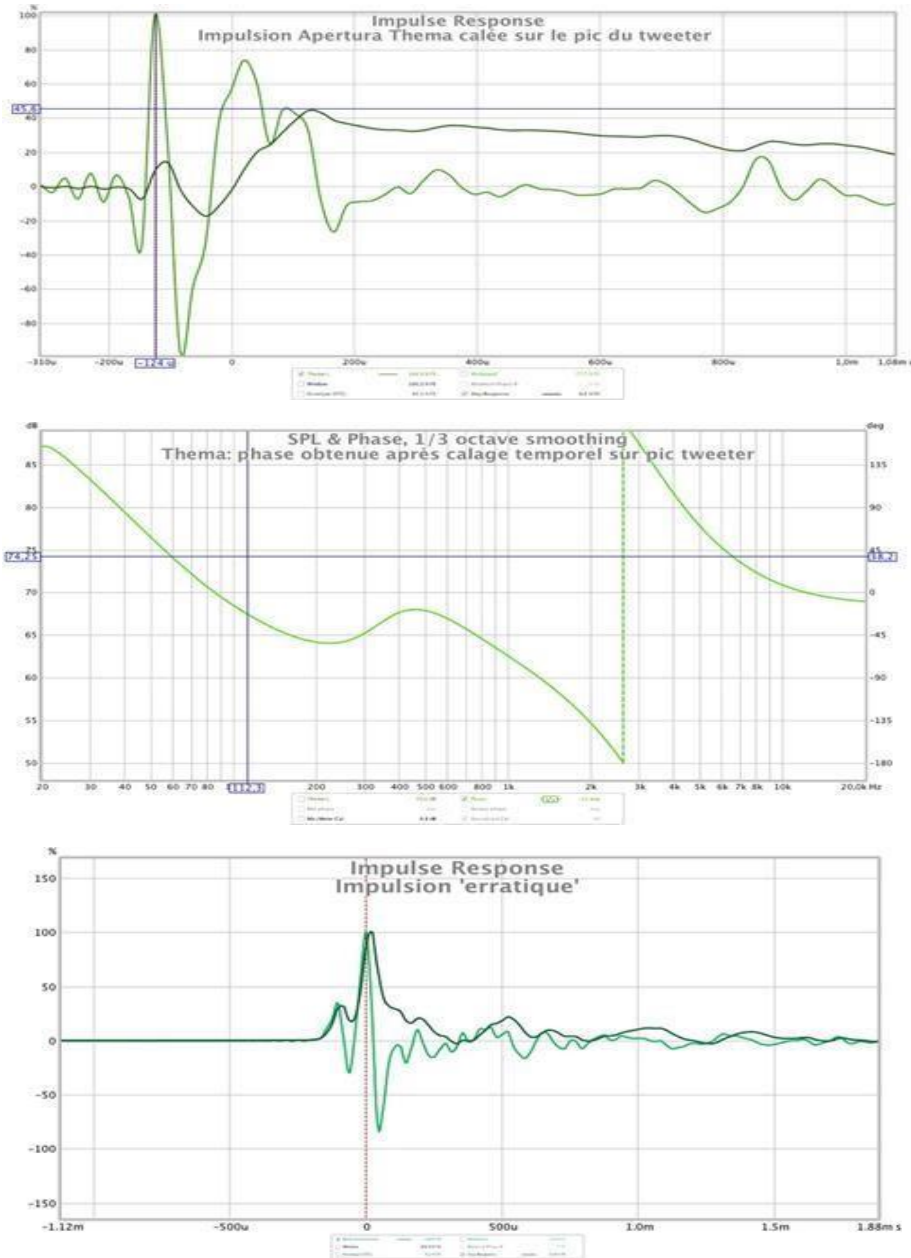
It is then necessary to eliminate as much as possible the reflections due to the local. This deserves a subject in itself and this is the subject of much debate.

Once the calibration done for the 2 speakers (right and left), we will export the measure REW of the phase in the RePhase software where we will actually correct the phase.

Note that it will be necessary, in a second time, also proceed to an export of the amplitude in RePhase to correct the amplitude.

These two operations are necessarily separated because the amplitude obtained with the windowing to obtain the phase, is unusable for defining the amplitude correction (because it eliminates the reflections that one needs for the amplitude).

Data export is done by selecting: 'File-> Export-> Measurement as text' in REW.
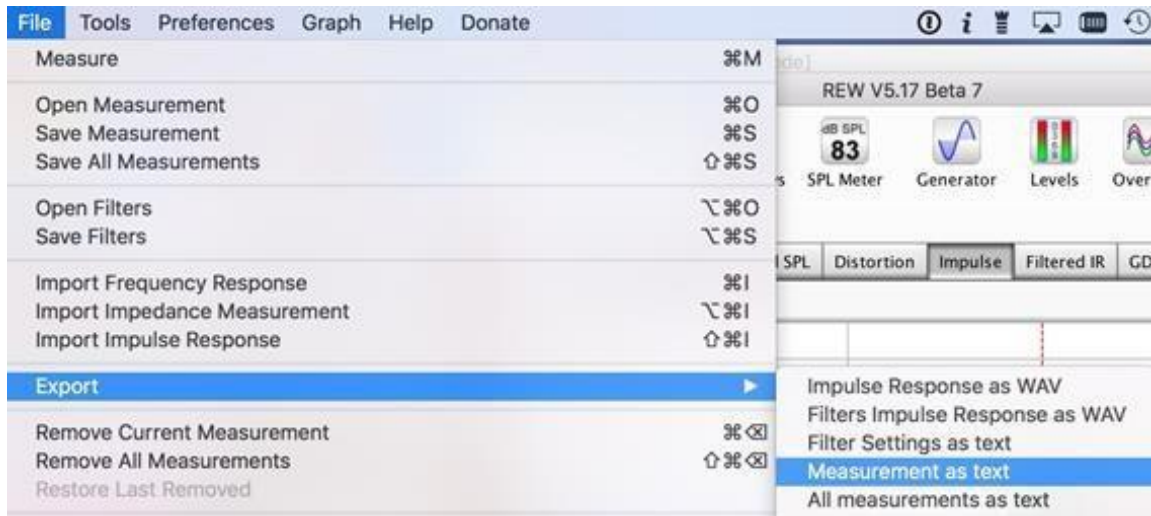
This gives a text file (readable with any text editor) for each speaker that can be named as desired.
Example: LeftEnglish.txt Phase and RightEnterprise.txt
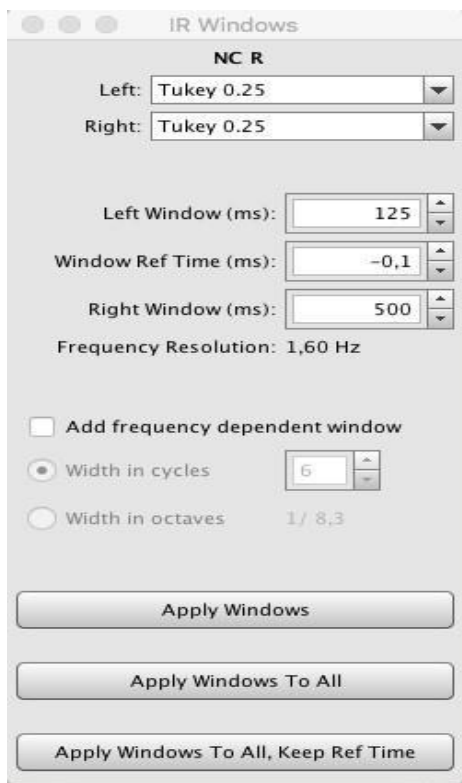
And while we are there, we will need to export the same data but for the amplitude this time.

To export the amplitude, you must first remove the windowing to find the full amplitude. Just uncheck the "Add frequency dependent window" box and click Apply Windows (on the Windows IR window).

Then, before exporting the data, one can choose a "relevant" smoothing to perform the amplitude correction. Everyone can decide which makes sense to him. The variable smoothing (Var Smoothing) proposed by REW seems interesting because it smoothes at 1/48 in the bass and rises gradually to 1/3 in the treble with a waypoint at 1/6 at 1000Hz.
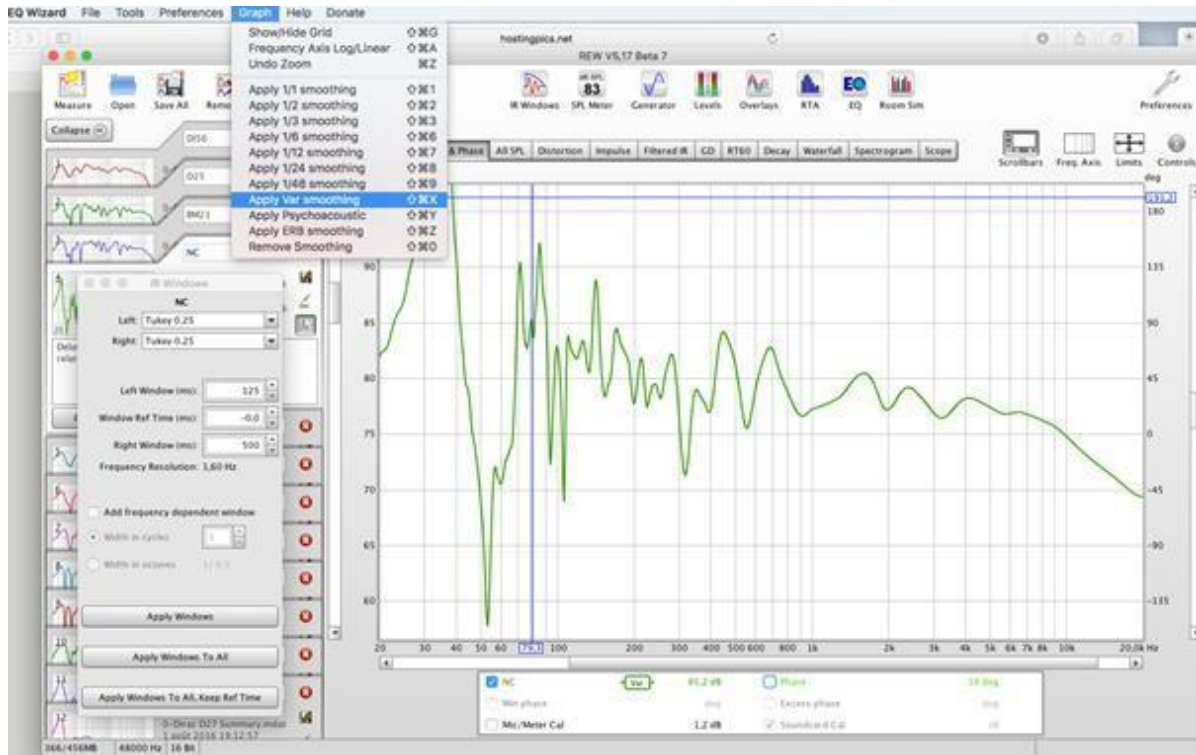
In any case, RePhase can test all possible configurations and so it is possible to evaluate the different listening results and to choose the best for oneself. Once the chosen smoothing applied to each of the speakers (or to the answer 2 speakers together), just export the measurement as for the phase by naming the files differently from those of the phase.

Example: Amplitude LeftHead.txt and AmplitudeHeadEnter.txt, or AmplitudeEnceintes.txt

4 Using RePhase for Phase and Amplitude Correction
4.1 Preamble

Once you export the REW measurements, you can launch the RePhase software (under Windows). *NB on the export REW: it is necessary to take "Measurement as text" in order to register only the current measurement (the one selected). Then give the file a name and then import it into RePhase.*

REW exports the data as it is displayed, that is, according to the smoothing used and possibly windowing. For the amplitude, you do not need windowing (and it is the opposite for the phase) and we can use the smoothing that we want. There is the Var Smoothing that is interesting but other options work too.

We are entering the correction phase itself and the objective here will be to correct first the phase of each channel, (ie of each speaker), and then correct, (if you wish, because this is not obligatory), the amplitude curve for the global response of your system is in line with your goals.

The operations will therefore begin with a first adjustment of the phase and then the amplitude response and will end with a check that the phase has not moved during the amplitude correction.

To do this, in RePhase, do File / Import Measurement ... or, more simply, do drag the first bar (for example LeftPrevious.txt) to the window RePhase.   RePhase will then display this measurement (note: it will surely be necessary to adjust the scales so that the displays are visible, for example by putting -80db in the box 'Gain offset' (on the Measurement tab) to calibrate the measurement that was taken in REW at 75 or 80dB, to the 0db RePhase, otherwise we see nothing).

Note: Data imported into RePhase (phase or amplitude) are only aids visuals to define the corrections that we will want to apply, but they do not enter in the calculations made. We can therefore manipulate the display scales and make them REW measurements at the desired level (dB), this has no impact on the calculations made by RePhase.

You should get a display like this:
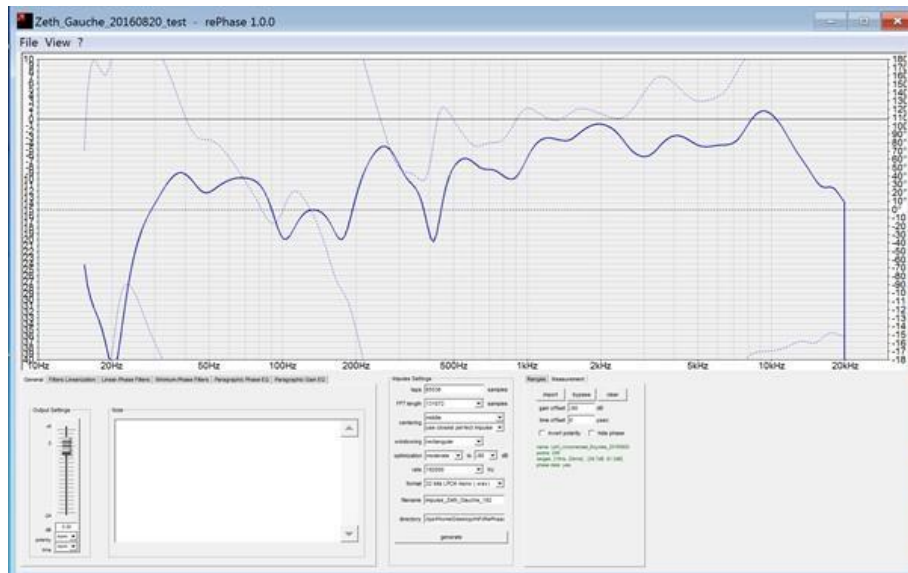
4.2 Correction of the phase
The first operation is to correct the phase.

It is very useful here (but not essential) to know the passive filters of your speakers (cutoff frequencies and slopes) and therefore do not hesitate to look at the documentations available!
If you know the passive filters used by your speakers, you can use this information and fill in the data in the 'Filter Linearization' tab. If not, you can try to guess by relying on where the phase changes more.

Here, a filter of 36db at 150Hz corrects the first rotation in the bass.
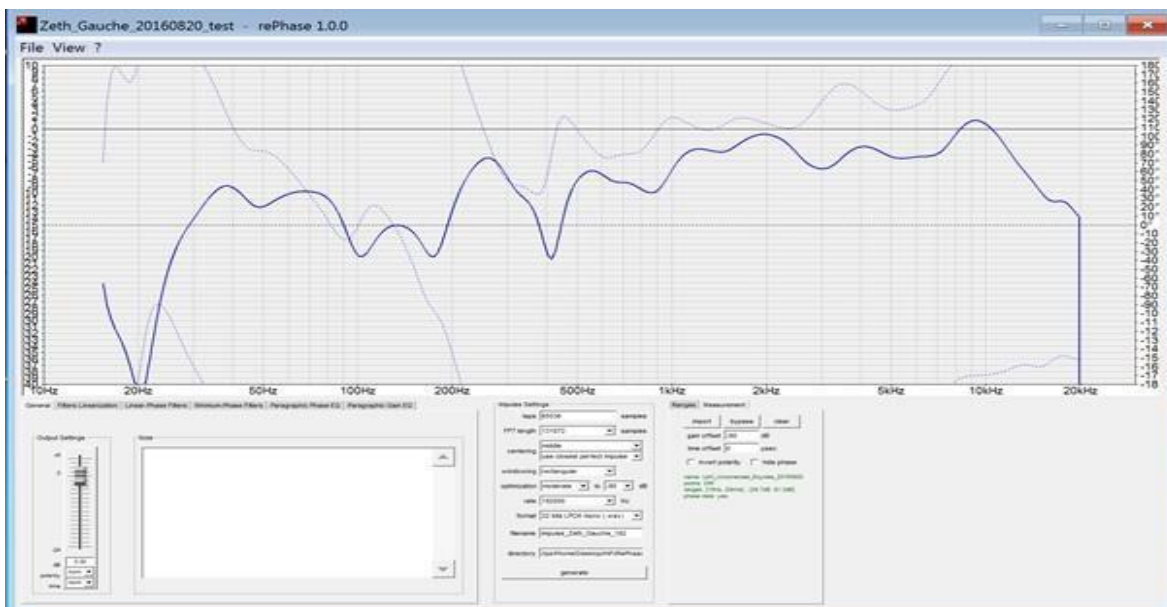
For example: on the Giya G3 (4-way + bass reflex vent), the cutoff frequencies are 220Hz, 880Hz, 3500Hz to 24dB of slope. So just enter these values so that the phase is already much more linear.

On Thema (2-way + bass reflex vent), the cutoff frequency is at 2,800Hz with a unknown slope, but by putting 24dB of slope, one realizes that the phase becomes very linear and that a simple adjustment can make it flat very easily.

Then you have to go to the 'Parametric Range EQ' tab, put the range at 180 degrees (for have more latitude to correct) and play with the sliders to finish adjusting the phase (with a little patience and usually it's easy enough). And you can fine-tune the settings using more detailed levels by adding 'Bank' and selecting 'Presets' at 1/3 octave, for example (but the Bank 1 is usually sufficient).

Finally, one can, at leisure, modify the precise frequency with which a cursor intervenes, as well as the intervention width by acting on the Q and Hz parameters.
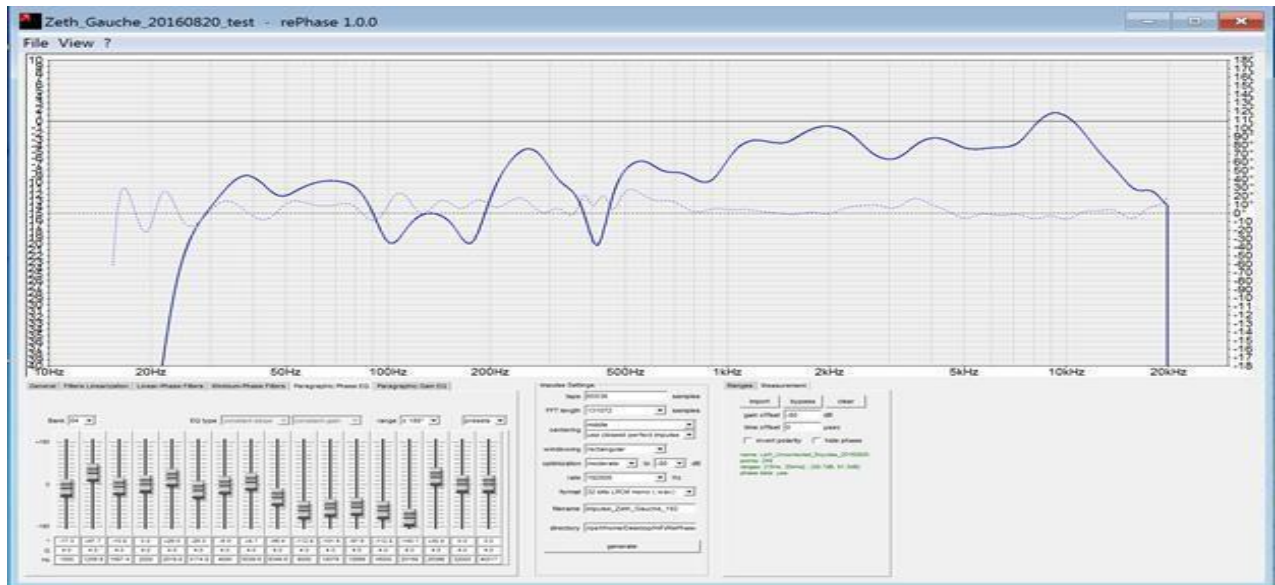
Here is what happens after a few operations. This must be done for each of the channels. The adjustment level of the phase is left the free choice of each. It is quite possible to reach a phase almost completely linear from 20 to 20,000Hz. Is it desirable and is it positive to listen? It will be for everyone to decide.

EDIT: Since writing the first version of this tutorial, it has been highlighted that a over-correction of the phase induces an important pre-ringing that could be audible and so embarrassing.
To get an idea of what is pre-ringing see the following post: Pre-Ringing And on his audibility: Pre-Ringing Audibility

The main recommendation is therefore to try, on the one hand, to get closer to the minimum, and not to obtain a flat phase at 0 °, and secondly, not to correct excessively the phase in the bass, the linearization with the filters being generally sufficient to obtain an excellent result.
Similarly, the objective being to correct the speaker itself (without the effect of reflections), it is better to have the same correction on the 2 speakers. We will be able to use independently of the 2 bars (right and left) to find the single correction adequate.

We will read on this subject with interest the tutorial realized by [Bear], whose interventions on the forum computer-audiophile, prompted the authors of RePhase and REW to integrate news facilitating communication between these 2 software and thus making the corrections much more efficient!

➢ **Method of measurement of [Bear] integrating the last functionalities of REW and RePhase integrated by their designers to make the job easier!**
• Download the following document: "Bear" method from 9 measurements

4.3 Amplitude correction

The phase settings are then dragged without touching, the measurement of the amplitude of the corresponding channel on the RePhase window. We see that the phase becomes erratic, but this is no longer the intention, because we will only be interested in amplitude without touching phase settings. (We can also remove the display by checking "hide phase" on the Measurement tab, which may make it easier to read the amplitude).
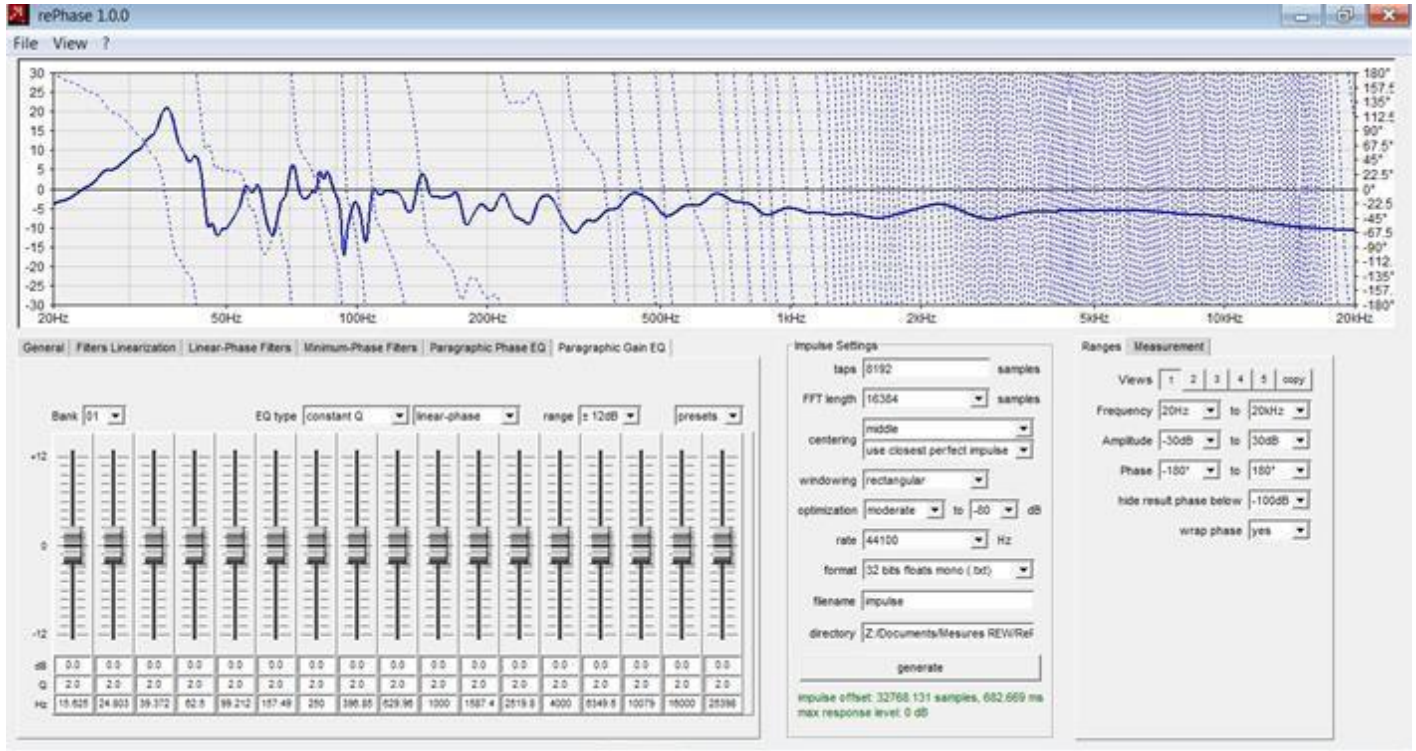
The goal here is to use the sliders on the 'Parametric Gain EQ' tab to change the curve according to his desires. Here is what we can achieve. You will notice that we have avoided use positive amplitude correction to avoid the risk of clipping by the following. It's a bias. You can also equalize by increasing the amplitude.

In this case (amplitude increase), RePhase indicates, when generating the impulse, the max response level (xx dB) below the button "Generate". We can decide to keep this setting because this is standard practice Dirac, Trinnov or Acourate, but in this case it is necessary to reduce the overall level in order to bring the maximum point at 0dB. If we do not do it, we will have digital clipping and it can be very audible!

It's very easy to do with RePhase, since you only need to apply a mitigation on the tab "General", of the desired value (see graph below with attenuation of -6 dB example).

For those who have scruples to lower the level, we can note that Dirac lowers it default of 8dB, Acourate of 10dB and Trinnov leaves the user the opportunity to decide the maximum level of positive correction, and therefore of lowering the necessary level. With RePhase, we are exactly in the same configuration as with Trinnov, we decide to how much dB we allow ourselves to correct in positive and we compensate the overall level as much as necessary, so maximum flexibility!

Example below with attenuation of -6dB:
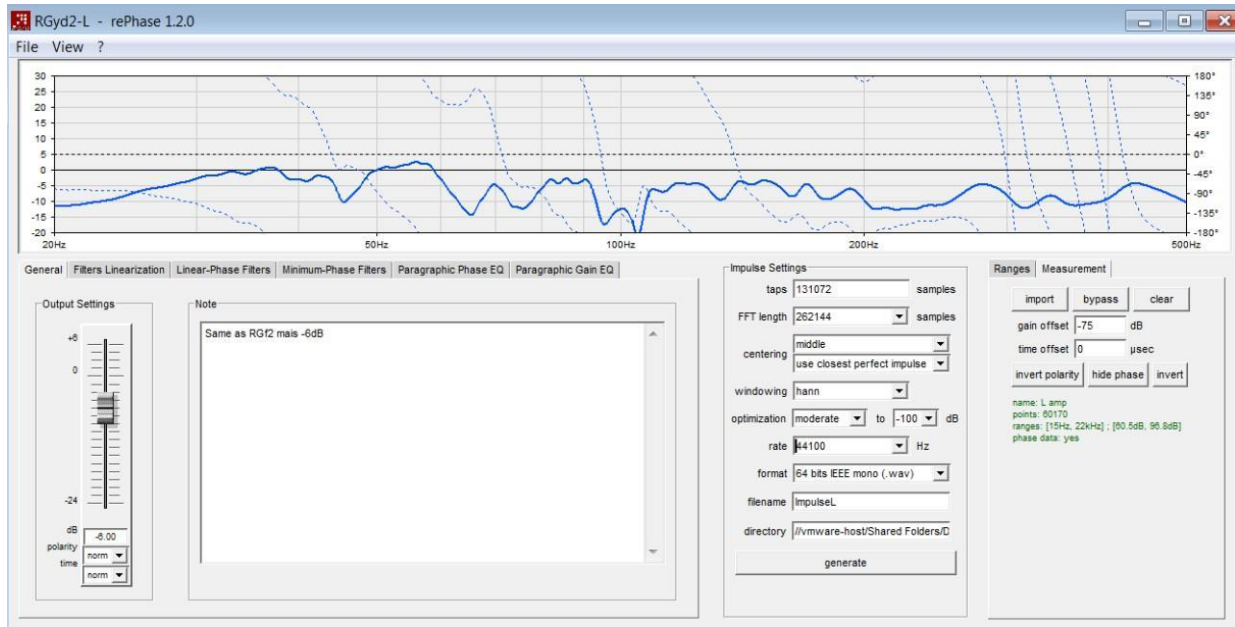
All this being said, in this area as in others, moderation seems to be the order of the day and we will avoid as much as possible to correct the impossible (the nodes (hollow) are often linked per piece and cannot be compensated, so let them alone) and we will limit as much as possible, corrections other than in the bass, except for possible gradually attenuate the treble with a descending target curve.

Note that do not hesitate to play with the different parameters including the Q that defines the action width of the correction (high Q = narrow action band and low Q = action band wide - Q can vary from 0 to as much as you want, but the really useful values are between About 1 and 25).

In addition, the RePhase equalizer is very flexible because you can choose the frequency of action in directly entering the desired value (which can also first be pinpointed precisely on the curve with the mouse cursor) and play the amplitude and the Q with the cursor (or moving keys, it's very convenient as well) and observe what's happening on the curve.

And since we can use several 'Banks', we have an equalizer with as many bands of frequency of action that one wishes, even if in practice only 1 Bank is largely sufficient most of the time.

Last point, one can choose on the tab "Paragraph Gain EQ" to keep the linear phase (Linear Phase) or let it vary according to the amplitude corrections (Minimum Phase).

Personally ([pda0]), I prefer to stay in "minimum phase" mode after adjusting the phase only according to the characteristics of the speakers.

This implies that the phase will be retouched with the modifications on the amplitude but note that in frequencies that are "minimum phase" (ie where there is no of anomalies generated by the reflections and the characteristics of the part), any modification of the amplitude to bring it closer to the theoretical ideal, will also correct the phase in the common sense (that is, bringing it closer to the theoretical ideal).

And conversely, the corrections that would be applied in non-minimum areas "phase" (that is, where there are anomalies related to the modes of the room for example), are not recommended (especially on the upside where they will not be effective) and we can not get a linear phase in this area anyway (except to generate a pre-ringing solid).

Anyone can, in any case, decide according to their priorities and the results to listening!

Below is an example of amplitude correction: in red the initial amplitude, in blue the amplitude as corrected by RePhase with equalizer corrections. Note that the correction was selected in 'minimum-phase', so corrections with impact on the phase.

The corrections include a positive correction of + 5.75dB, and with a general correction of -6dB on the "General" tab, we get a max response level at -0.25dB, as indicated by RePhase.
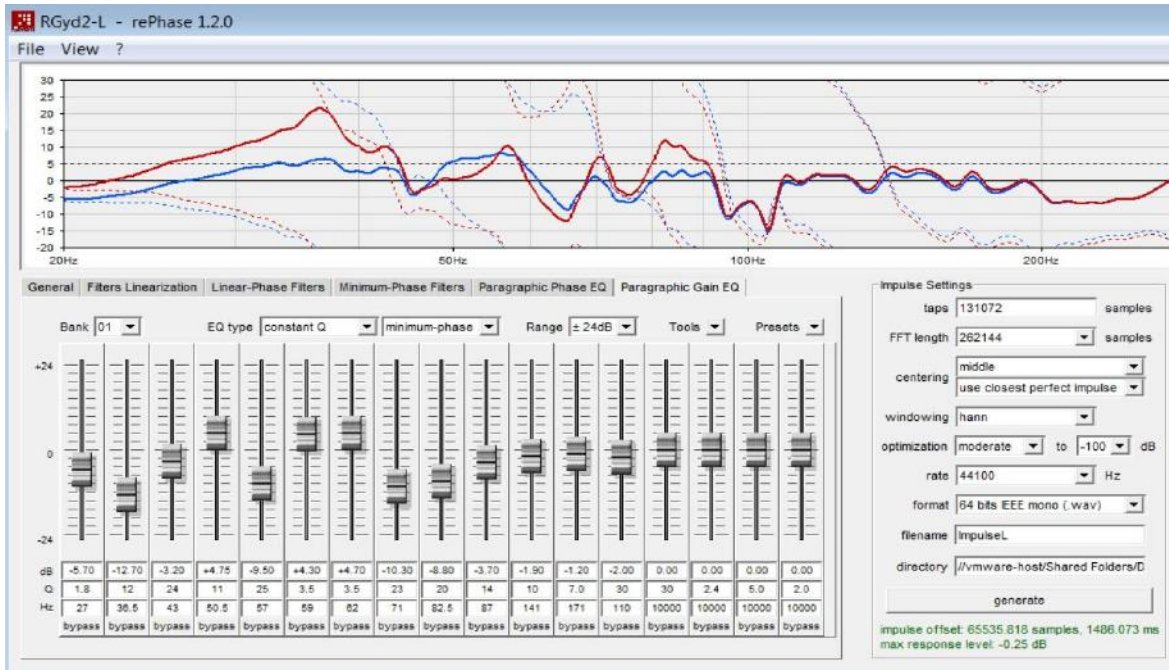
4.4 Generating Impulse Files

The time has come to generate **correction pulses**.

This is done by choosing the appropriate parameters in the RePhase window. We recommend **65536 (or even 131072) taps** (for those who have no constraint of processing power - that is, anyone who will use a PC or Mac mini or Macbook) and **64bit IEEE mono (".wav")** as the pulse format.
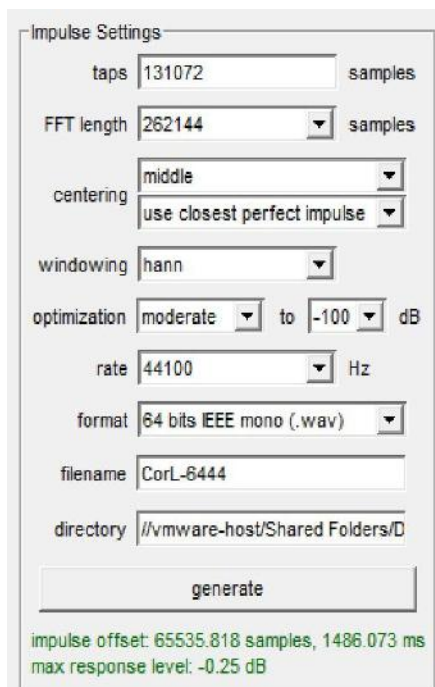
It is necessary to generate an impulse for each sampling frequency that we uses for each of the channels.

Note that RePhase gives you, at the end of generation, the indication of the maximum level of correction used. If the level indicated is less than 0db, you will be able to do without anti-clipping protection in JRiver.

For the initial tests, it may be more efficient / fast to be limited to generating the pulses in "32-bit mono LPCM" and 48kHz, because it's the only impetus that allows you to realize simulations and thus validate the options chosen with REW.

And once the result is satisfactory to the simulation, it will remain to generate the impulses of correction for all configurations used (in practice 44.1 / 48 / 88.2 and 96 should be suffice in the vast majority of cases, but it is possible to generate 176/192 and more if need was ...). For those impulses that will actually be used in the reader, we will generate them in "64bit IEEE mono" in order to have the best quality available for calculations of convolution in the reader.

5 Simulations of the results with REW

For having performed a few dozen simulations and post-simulation measurements, we can confirm that the simulation with REW produces an absolutely remarkable result and almost identical (even with smoothing at 1 / 48th of an octave) to that obtained with the effective measurement of the result via convolution of JRiver or other reader.

It is therefore perfectly reasonable to proceed with the simulations until we obtain a satisfactory and not listening (and possibly measuring) versions retained, but here too, everyone can do as he sees fit.

To perform a simulation, it must
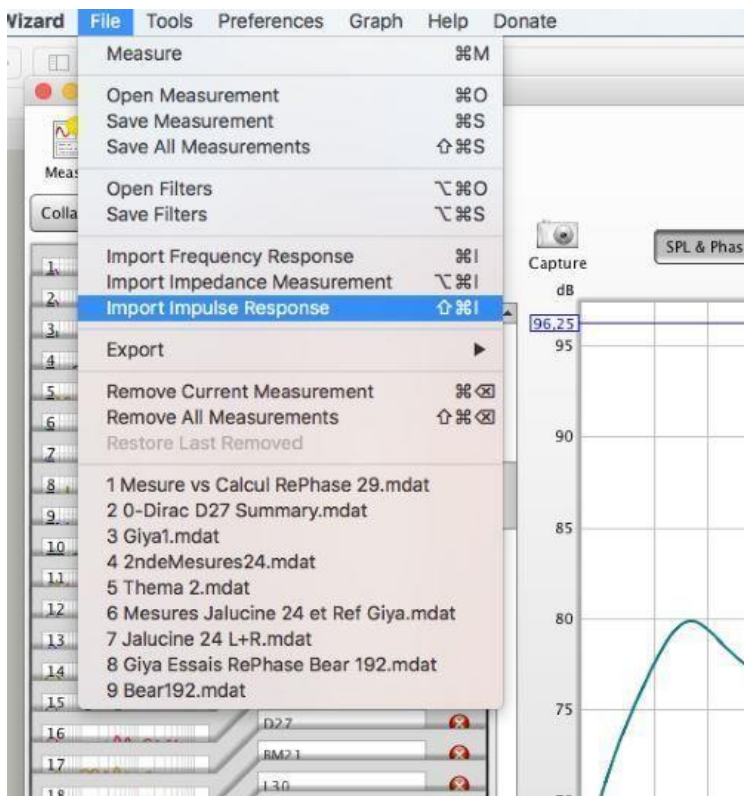
All first import the pulse generated by RePhase in REW, using the menu File / Import Impulse Response.

The impulse is a ".wav" file and is imported as is in REW (attention: it must be at "32bits LPCM mono" format and 48kHz (if the measurements were at 48kHz, which is the setting recommended) to be able to be read by REW).

Then we will "multiply" this impulse with that of the measurement of the considered path (left pulse with measurement of the left speaker, for example).

Warning: no check that multiplies many compatible things is done. So if we make a wrong file, we get sometimes funny and incoherent results always.

To perform this multiplication, simply place on the window measuring the track considered (let's say the left here), then click on the 'All SPL' button in the window REW display.

Then we select the 2 curves that we want to multiply (here the measure of the left channel   and the RePhase pulse of the left channel just imported).

Note: the measurement of the left channel must be without windowing, otherwise the operation will be incorrect. So make sure that the windowing is removed (box unchecked on the IR window). We choose the operation A * B and press the generate button.
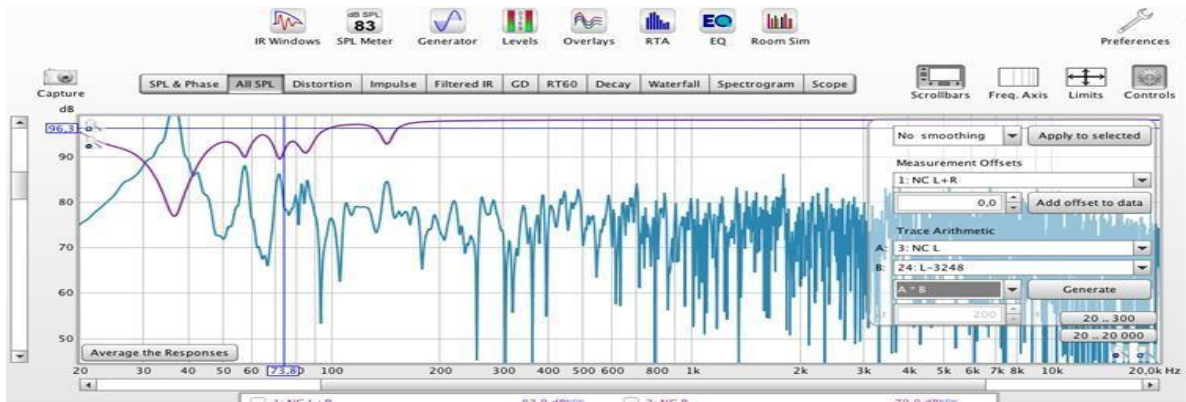
A new measurement then appears, and we can go and observe it.
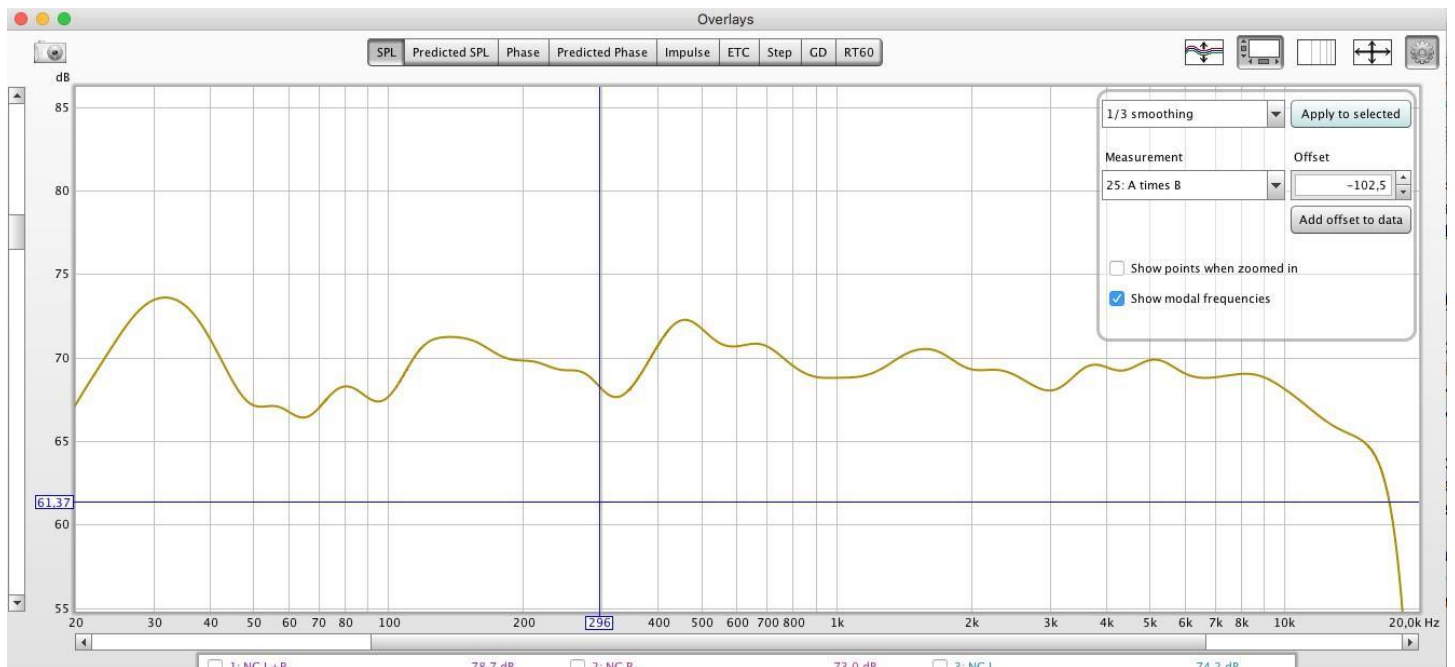
For some unknown reason (perhaps a REW bug ...), the simulated curve is much higher than initial measurements (from 100 to 150dB sometimes). So he goes First of all, you have to bring your level back to the same level as the amplitude of the track measurement. left without correction, in order to be able to compare them.

The easiest way to do this is to go to the Overlays window, and display only the measuring the left channel, and the calculated measure A * B, with a 1/3 octave smoothing. We see that the measure A * B is here 102dB above the measurement of the left track and it is necessary therefore bring it back to the same level to facilitate comparisons. It must be selected in the "settings" menu of the overlay window, then offset -100db, then manually adjust until the 2 measurements are at a level comparable (say aligned at 1000Hz for example). Then click on 'add offset to data', to complete the simulation calculation.

It is convenient to rename A * B to 'Corrected Left Path' for example. In fact, a good coding is recommended because it is unlikely to have a perfect result at first shot and without numbering and a minimum of organization, we are quickly lost ...
For the correction of Giya, [pda0] performed more than 30 iterations before having a result close to that of Trinnov or Dirac (but hey, a lot of wasted time trial and error software learning). With a little method, we can achieve a very good result in 3 or 4 iterations.
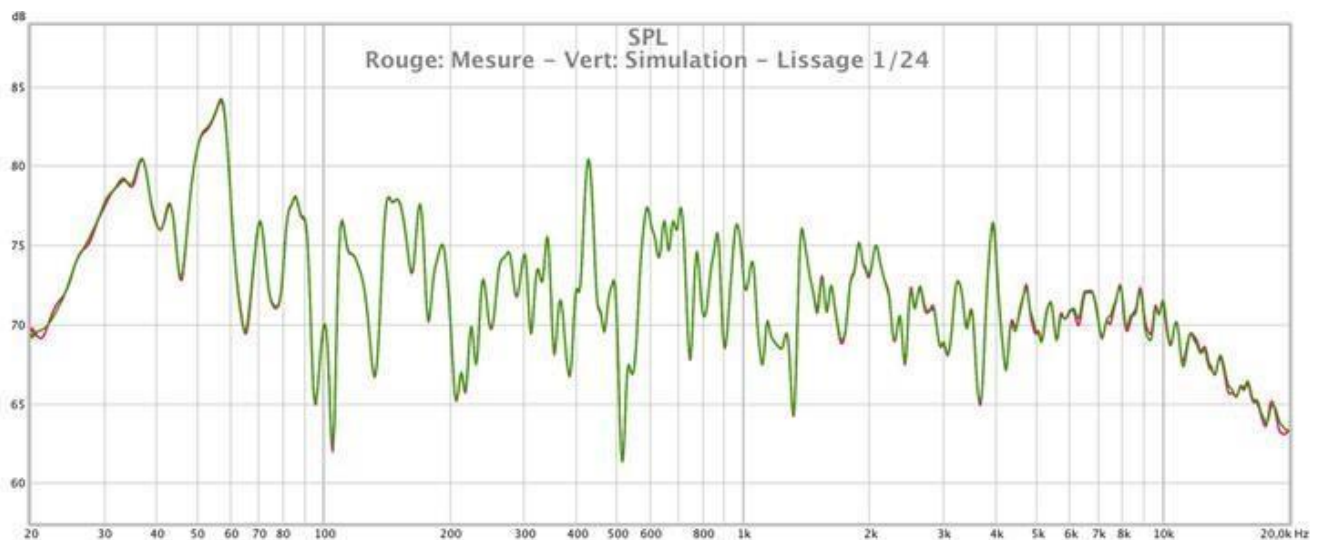
Once this operation is done for the 2 channels (right and left), we can simulate the behavior of the response with the 2 channels in operation, simply by the addition of the two simulations (A + B instead of A * B).
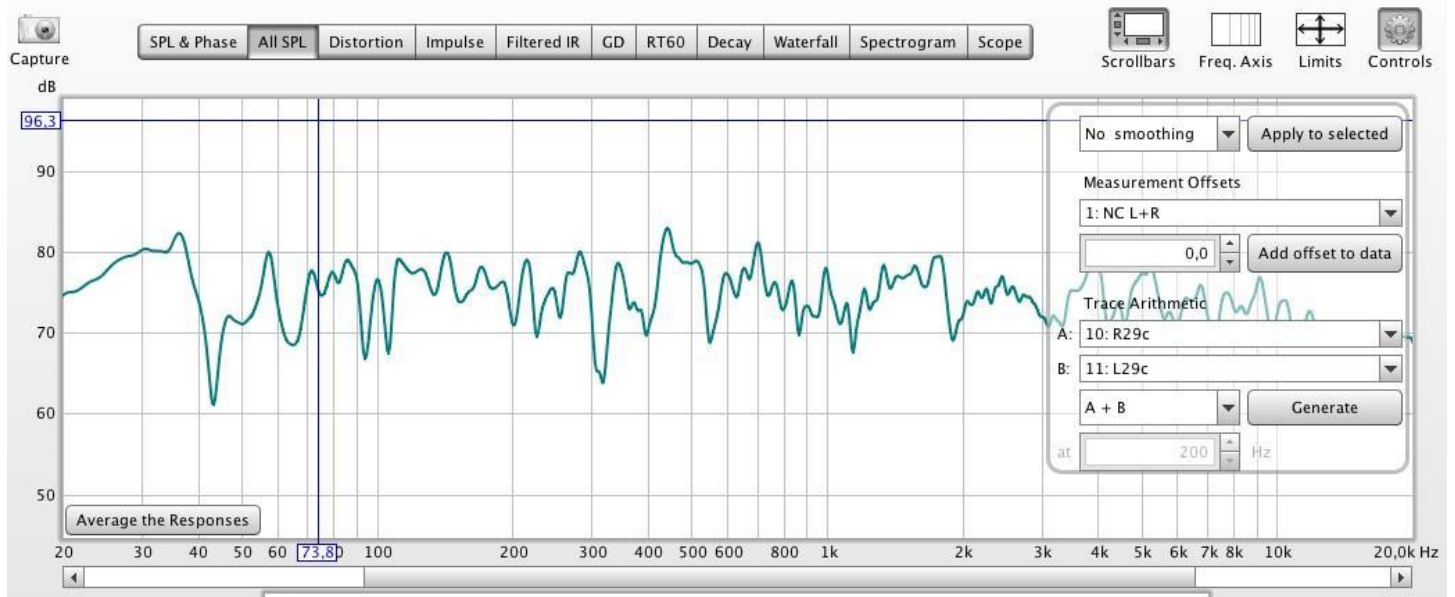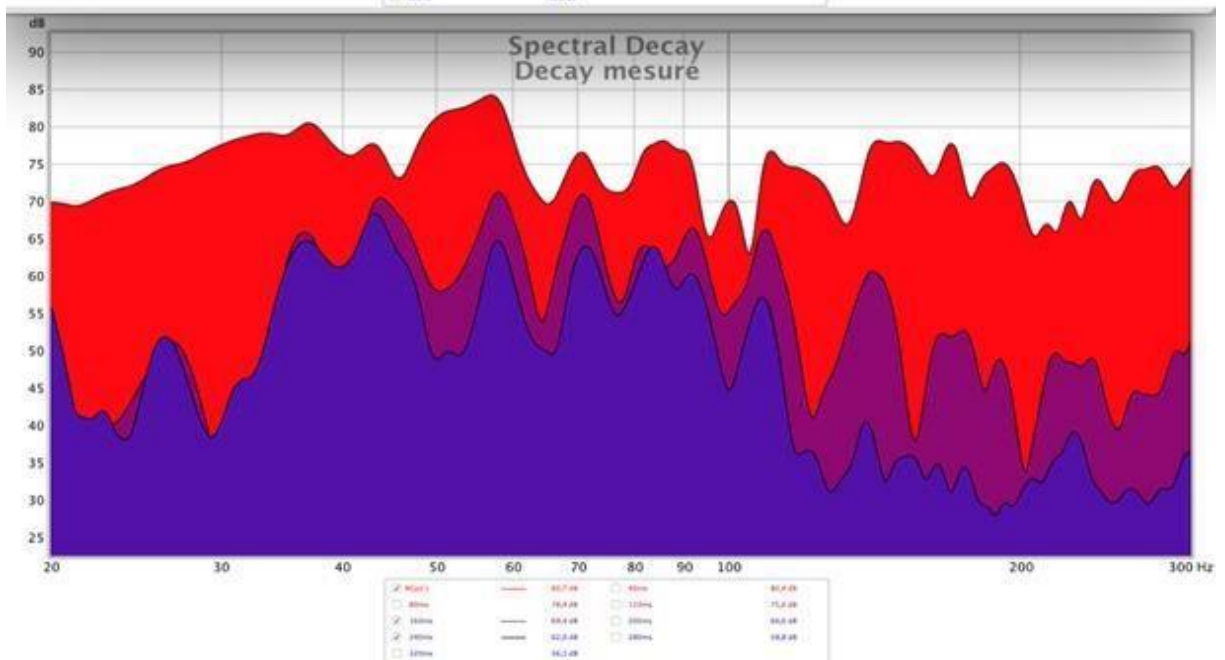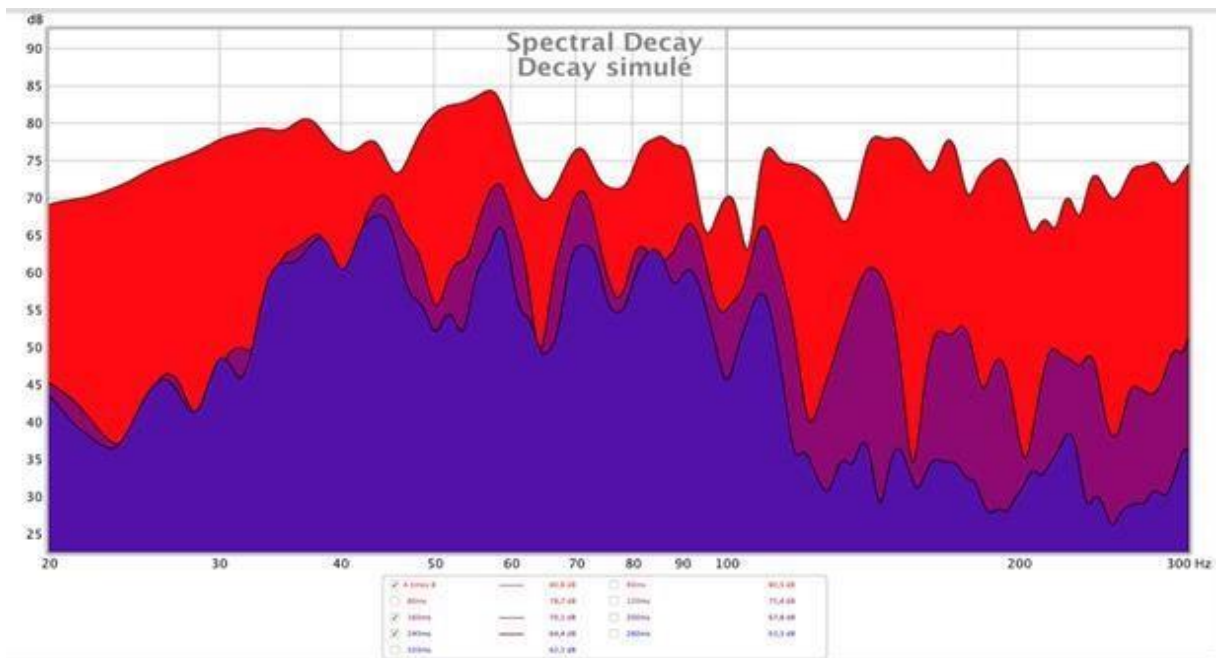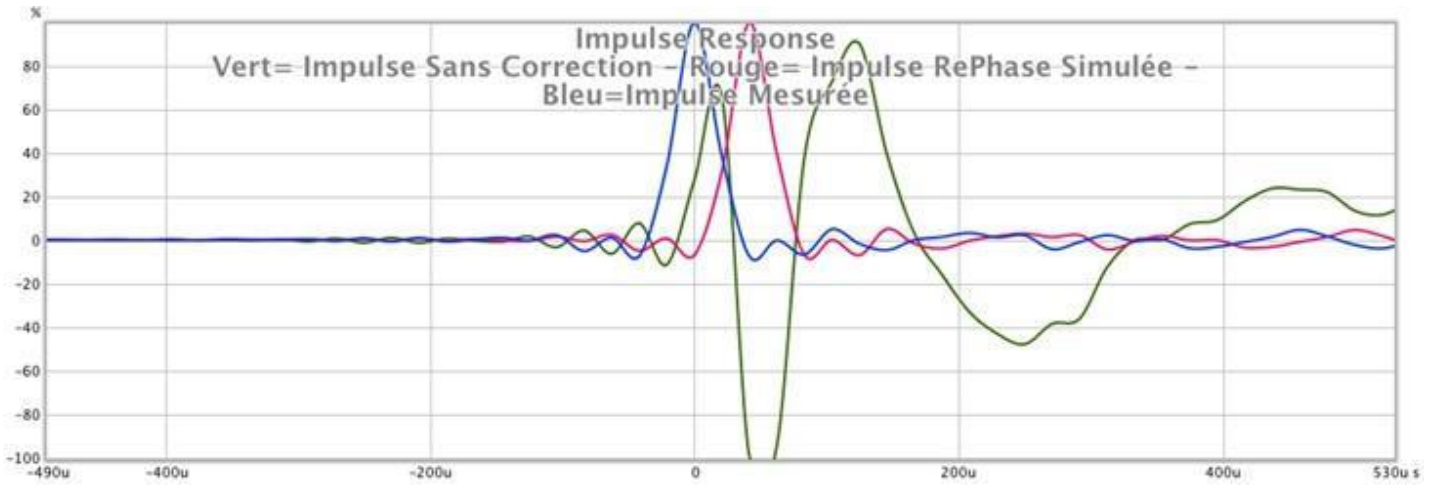

This operation is very interesting and informative because we can see in real size the effect of the phase on the amplitude. The addition of the 2 channels is not a simple addition because if at a given frequency the 2 speakers are in opposition of phase, it will be a subtraction! We can easily observe with RePhase that the same amplitude on the left track for example but with an inverted phase at certain frequencies (easy to simulate with RePhase) produces a totally different result when doing the addition with the right lane. And the most "impressive" is that the measurement perfectly confirms this behavior!

We can also note that if the simulation is exact for the amplitude (and this, whatever the smoothing), as well as for pulse and phase; she is a little optimistic for the decay, the waterfall, and especially the spectrogram. But it's not bad, and very useful to refine the settings without having to remeasure a lot of intermediate versions!

Comparison examples Simulation vs. Measurement (Amplitude, Pulse, Decay, Spectro):

Impulse Response
Vert= Impulse Sans Correction – Rouge= Impulse RePhase Simulée – Bleu=Impulse Mesurée

Spectral Decay
Decay simulé

Spectral Decay
Decay mesure
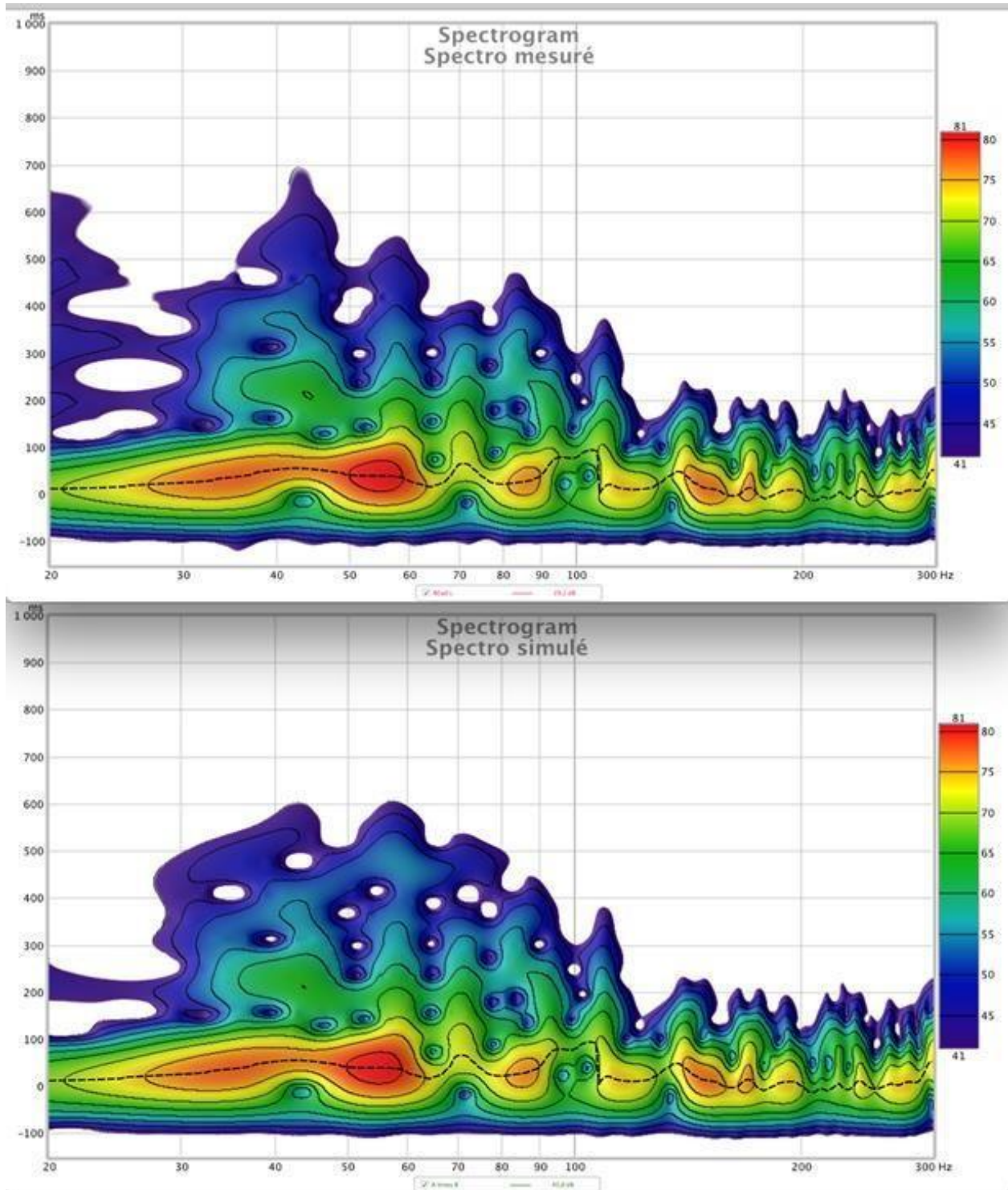
The simulations are extremely efficient and accurate, and we can rely on them to develop the settings you want to get before listening to them, and to measure eventually.
In the end, we realize that RePhase can play with all the parameters, and the Adjustment possibilities are immense and unequaled flexibility!

To have a lot done with Trinnov and Dirac, RePhase is, by far, much more flexible, but it's without a net, so all the nonsense is also allowed ... but it's also as we learn!

6 JRiver Configuration

Here is not the place to present JRiver in detail, and the links given above, as well as the huge literature available on the internet for this software should be enough to understand the interest of this software.

We use it as a dematerialized file reader, and just activate the function of convolution to use it with RePhase. All those who already use JRiver are already equipped to implement RePhase without changing anything of their habits.

To use RePhase, simply follow the procedure below which will consist of indicating to JRiver where are the impulses that were generated previously, and how wish he uses them.

To apply the corrections in JRiver, there are two possible strategies:

• either dynamically re-sample all your files with JRiver, for example in 24b / 192kHz if your DAC supports it, using the function available for this to do:
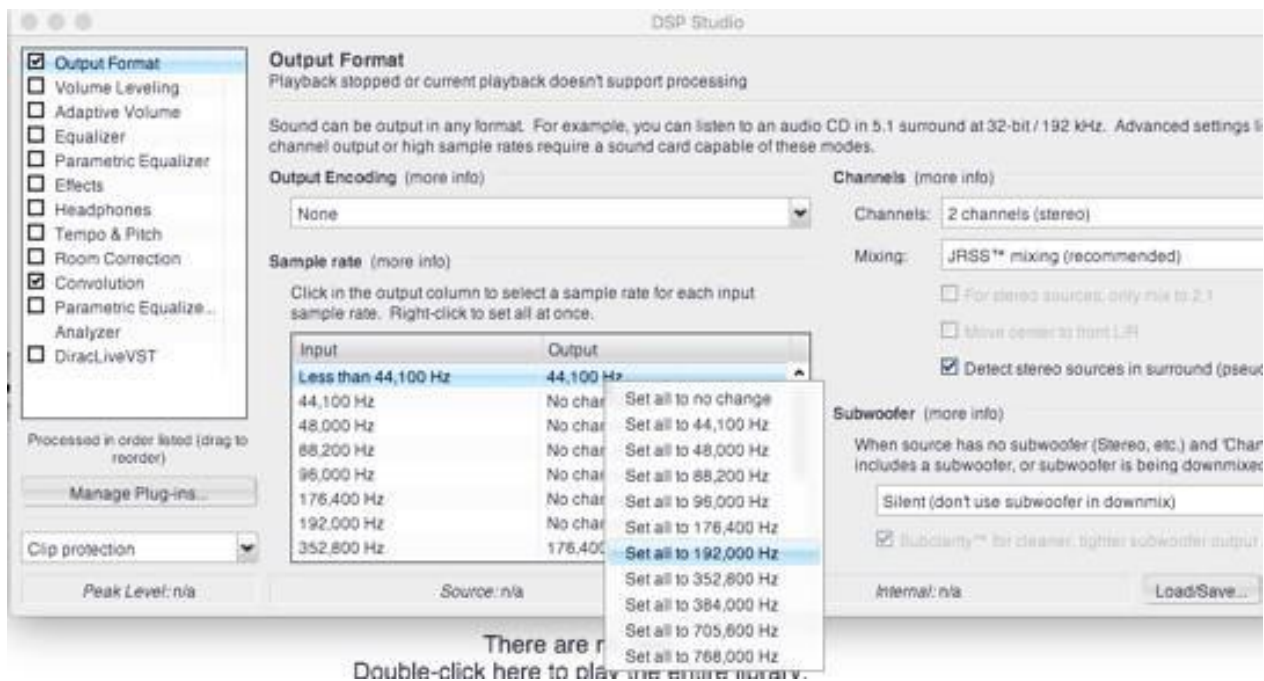
• In this case, you will simply need to assemble the impulses for each channel (so left and right for a stereo system) and assemble them into a file of configuration which will be called '2.0_192.cfg' which is an example:

• In this case, you will have to select the configuration file in the tab convolution of JRiver.
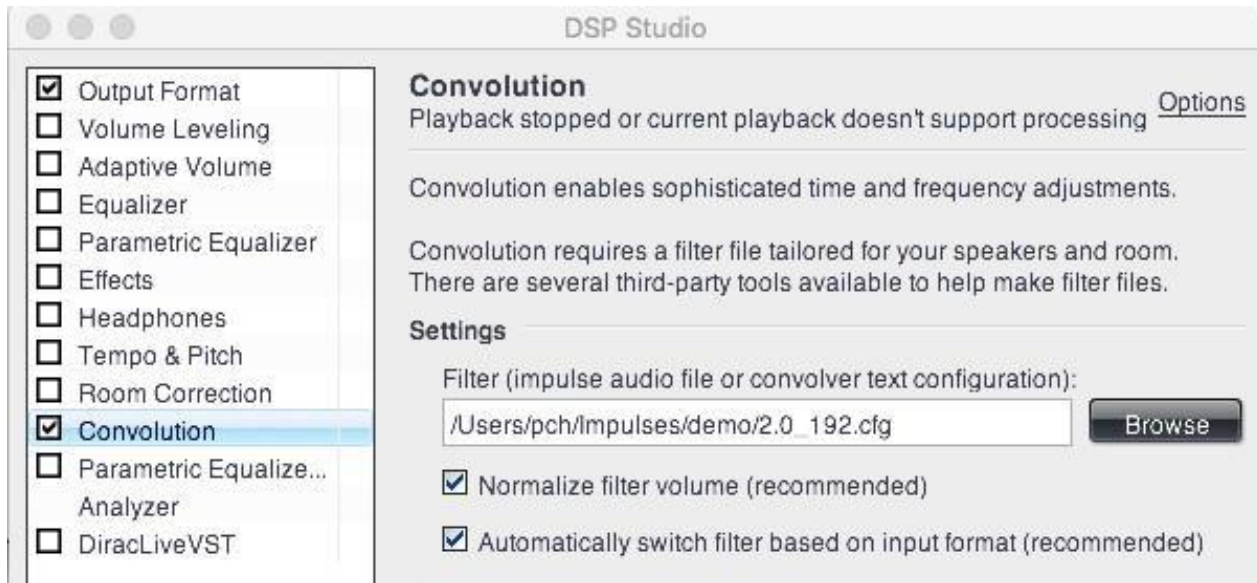
• you do not want to resample your files and want to keep them initial sampling frequency. You will then need to generate an impulse in RePhase for each channel (left-right) and each sampling frequency.

• For each frequency, you will need to write a configuration file on the model below. above, simply changing the frequency and the name of the pulses. Here by example a file for the 44.1kHz:
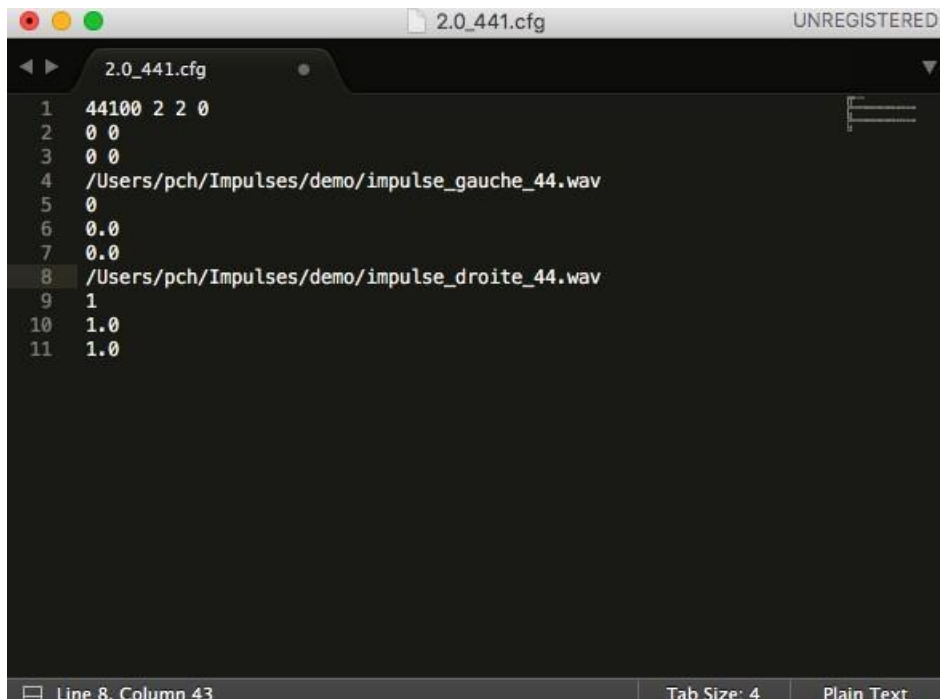
• Simply point to JRiver convolution tab to one of these files configuration so that the file naming rules are standardized, JRiver associates the right configuration file with each sample rate and select the right impulses:
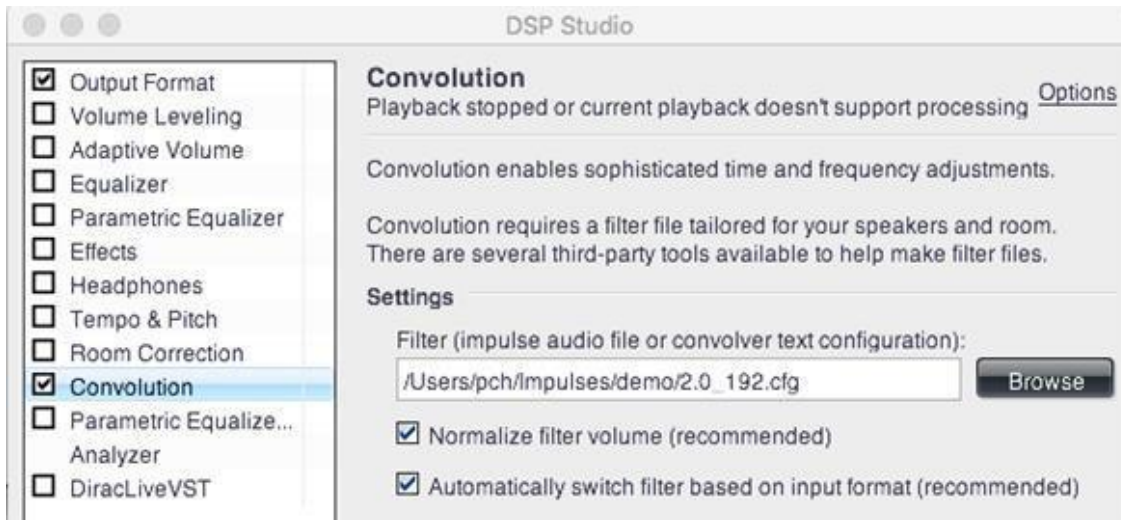
• Here are what the filenames in question look like:
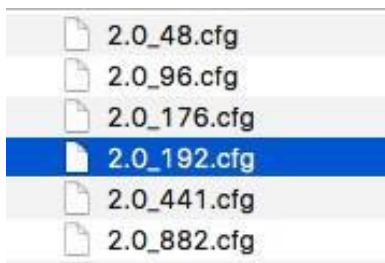
To facilitate the work of those who have arrived at this stage, we join on Dropbox (link attached) a configuration file kit for Windows and Mac, to simplify the creating these files.

You just need to change the settings in these files (path and names of impulse files) according to your configuration to use them, and to tell JRiver where to find these configuration files.
Windows files
Mac files

JRiver also has the advantage of being able to handle convolution and send the file corrected via DLNA on a network drive. This allows you to isolate the drive from the server, and to use a minimal and optimized reader close to the DAC and to let the "big" server hi-fi system to avoid possible electromagnetic pollution ...

The procedure to activate this function, which is a native function, of JRiver is described on the post of [Bear] below:
DLNA configuration

Note that this procedure is totally independent of the numerical correction, but it also works with the correction enabled, which is a more certain for those who use a DLNA player with a separate server.

For the followers of Roon, the procedure of implementation of the digital correction is very similar in principle, and very easy to use.

Below 2 screens that show the principle of implementation with Roon. The documentation is very well done elsewhere (but in English).

The first screen shows how to add the convolution filter in the DSP.

The second screen shows how to include the filter (just click on browse and go look for the configuration file containing the numerical correction as with JRiver).
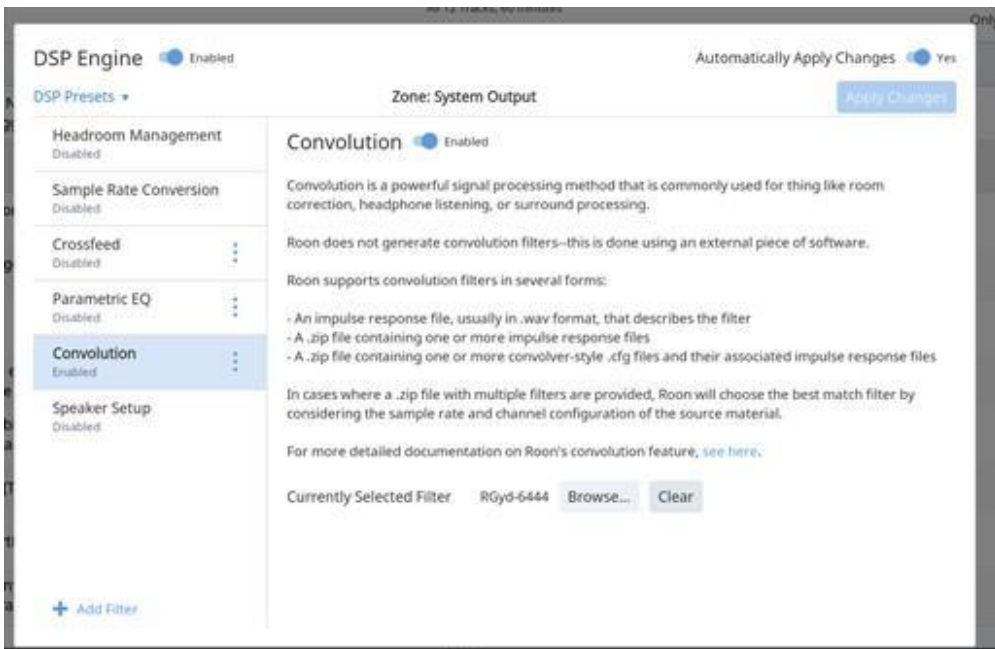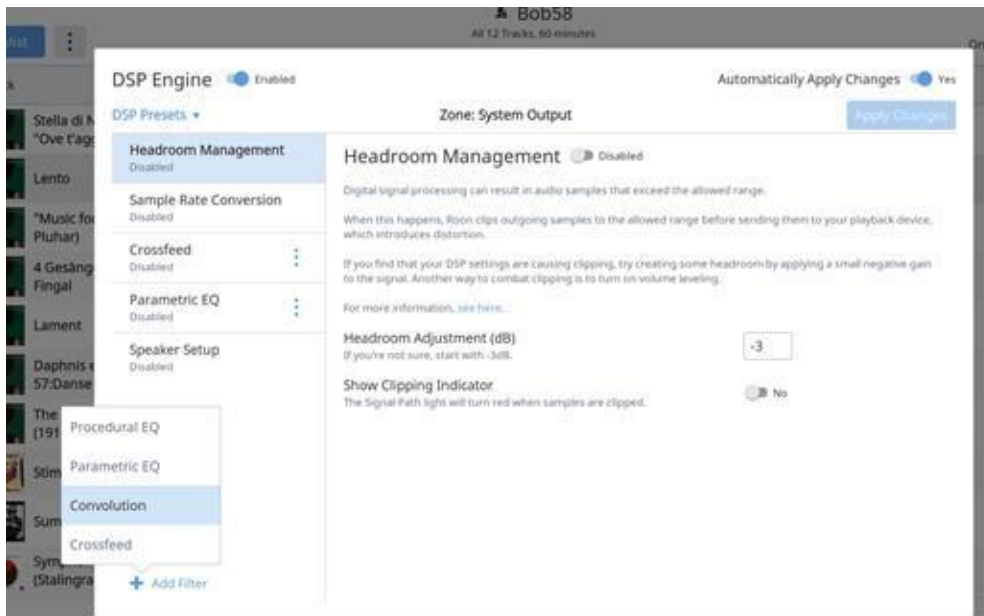
7 Measure the result with REW

This operation is not absolutely necessary given the quality of the simulation that REW product. Nevertheless for the curious or skeptics), it is possible to measure the effective result using the "indirect" method that allows REW to measure a impulse generated by the reader with convolution activated (JRiver, Roon, Foobar ...).

This procedure is described here: [Measurement tutorial with network reader or convolution](#)

8 Conclusions

Beyond measurements and other simulations, the only thing that matters is of course the result obtained listening!
And in a nutshell, the quality achieved is very close to that obtained with Dirac or Trinnov, not to say that it is identical ...

To measure is, in any case, an obviousness; listening, everyone will decide what the best, but for us, digital correction image and definition improvement that is indisputable, and this allows to adjust from very effective way the serious problems of resonances in the serious.

Technically, with these 3 programs, we have all the necessary tools to obtain an excellent result, subject to a little patience and method! Free admission is paid here through a relative complexity of implementation.

It is also by playing with RePhase, REW and JRiver (or other) that one realizes the work done by Trinnov or Dirac, who manage to automatically generate a result consistent and excellent without any information about the system and its environment!

Nevertheless, RePhase allows the audiophile (very slightly tinged geek) who knows his system and its environment to decide which corrections will be applied and which will not be with exceptional flexibility and precision!

In short, a product to try, at the risk of having to adopt it ...

9 Annexes

9.1 Supplements
9.1.1 Creating a Minimum Phase Correction Pulse in REW / RePhase

➢ **Tutorial realized by [Audyart]** :
(for those who want to download it: http://forums.melaudia.net/attachment.php?aid=16510)

**Minimum phase equalization with REW and RePhase**
"A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a frequency response A given amplitude curve is the limit optimization of the transient response of a system and its amortization before creation of pre-ringing, target that is found practiced with commercial software like Dirac, Trinnov or Acourate.

It can be done by successive iterations or by using the mathematical functions available in REW to extract the excess phase, with fairly good accuracy.

The initial measurement used for illustration is a two-way hr, filtered in QO windowing fdw 7 cycles + weighting 1/12 oct.
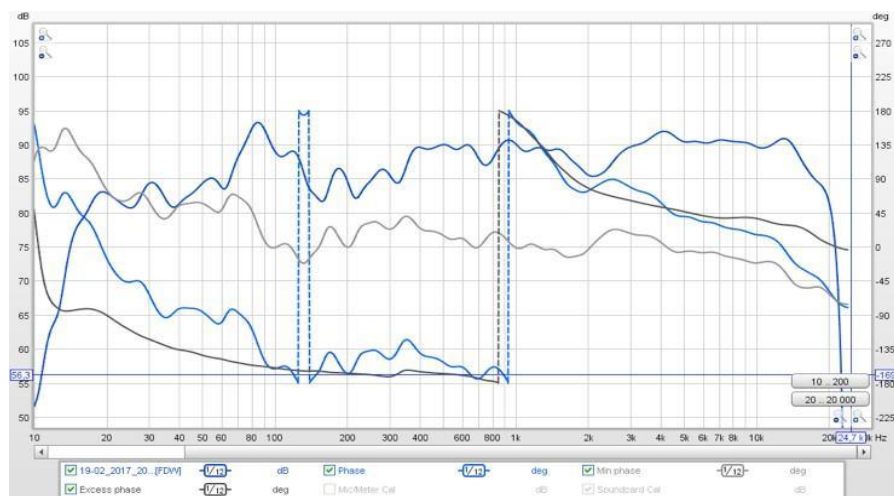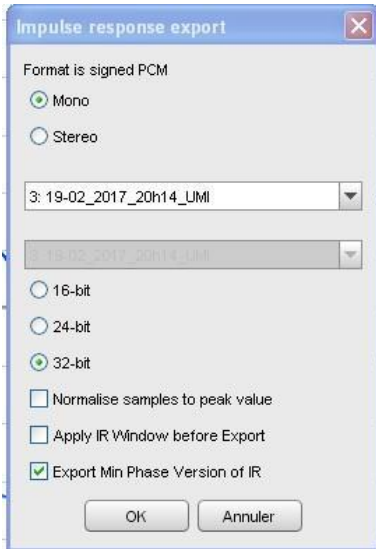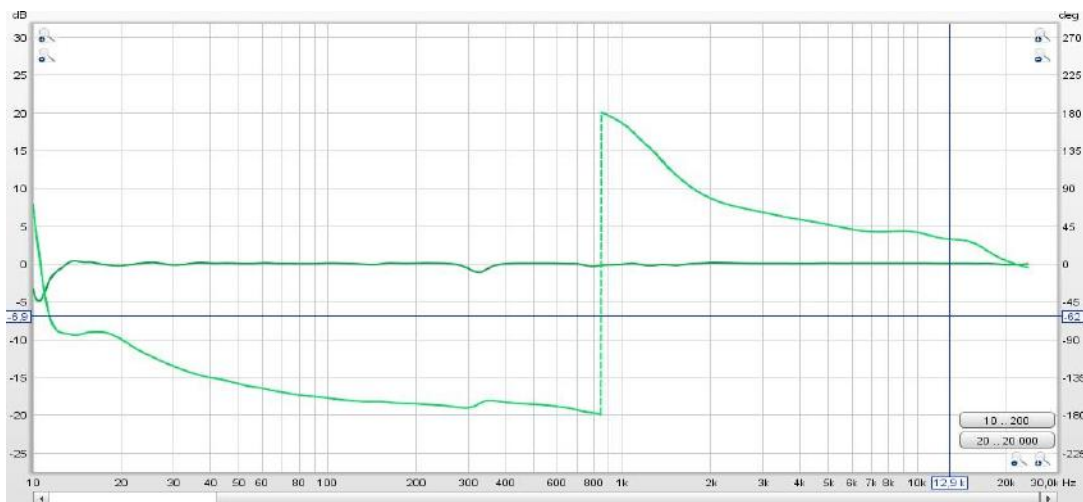
Export of the minimum phase pulse

And re-import into REW, aligning levels if necessary.

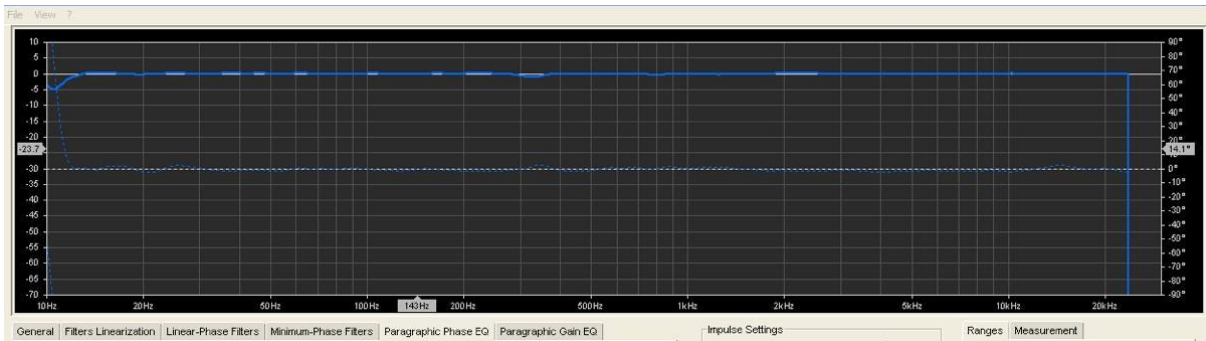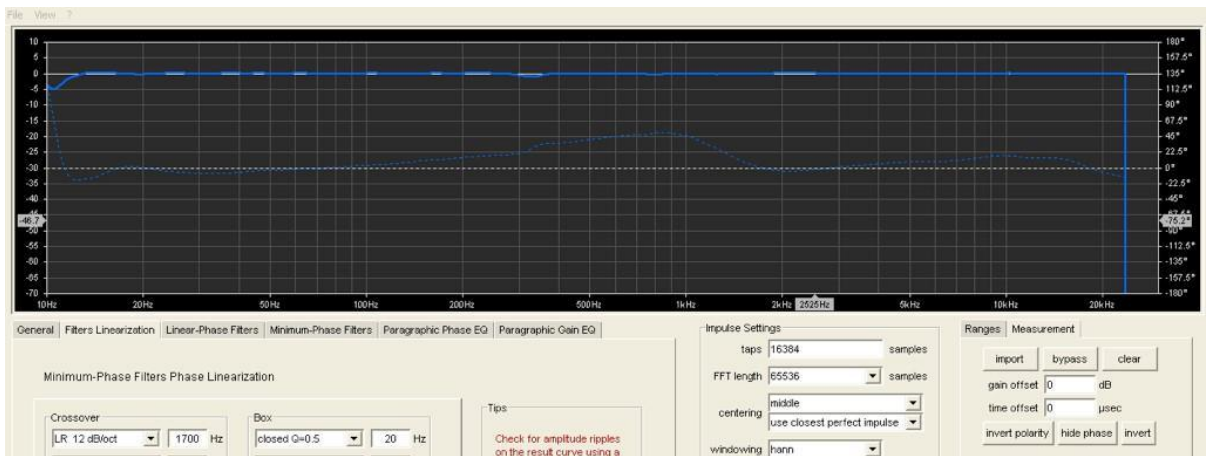Using the A / B function in REW we get an amplitude (almost right at 0 dB) and the excess phase:
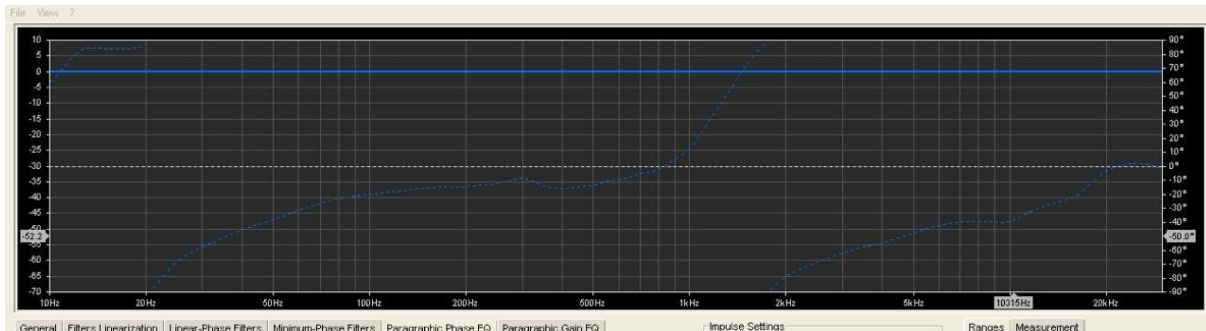
which is exported in .txt to RePhase:
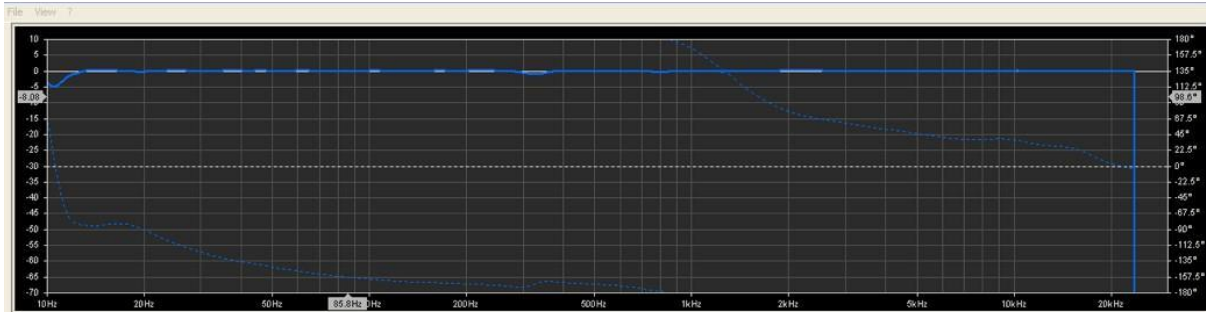
The phase curve is equalized via the "linearization filters" functions then the "paragraphic EQ phase "
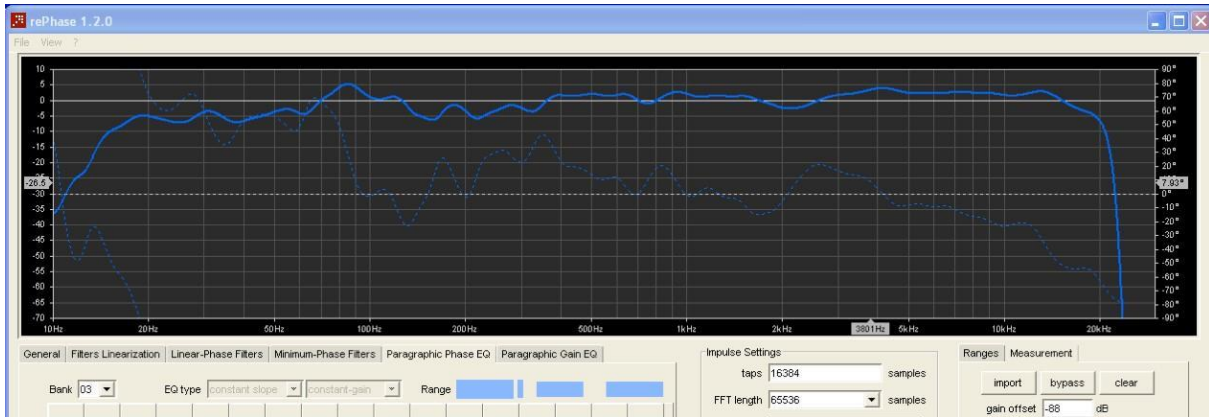

Then "clear measurement" to reveal the effective phase correction that is the excess reversed phase of the initial measurement.
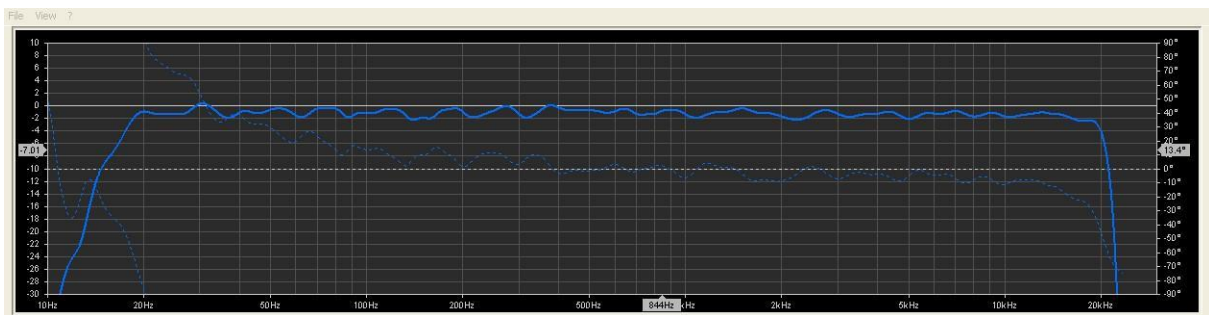

Then drag (export txt) the true initial measure A in RePhase:

and equalize it only with minimal phase EQ

The final correction pulse is created with the sampling of the measurement and dragged into REW to simulate the result with the function A * B

and observe the superposition between the obtained phase and the minimum phase, which do not diverge only a little at the ends of the band:

(the calculation of the minimum phase by the softs not being done there any way quite reliably)
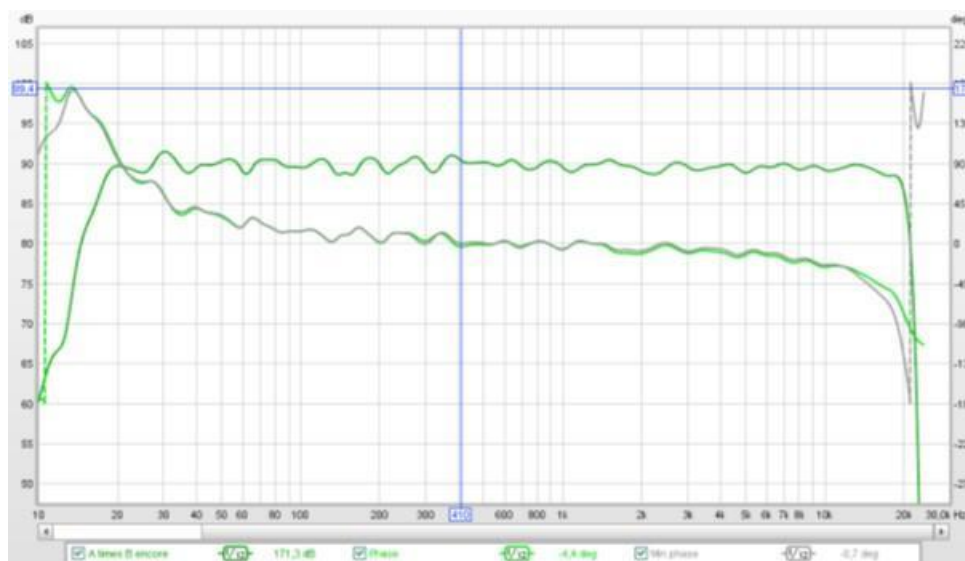
leaving only a very slight pre-answer to the step and showing a very good damping:
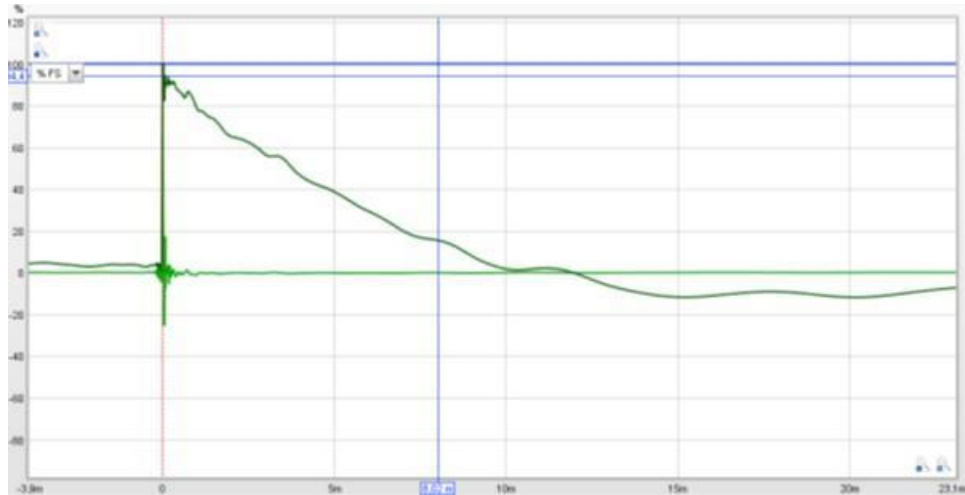
9.1.2 Additions to the measurement method proposed by [Bear]:

[Audyart] recommends in his tutorial to practice temporal windowing (7 cycles). This method has the small disadvantage of removing information.

- another method may be to make the correction on the basis of an average of 8 or 9 measurements as indicated above. This averaging method does not remove of information.

Indeed, with the different methods explained above:

- possible use of averages of measures to avoid having to make choices of temporal windowing (9-point measurements , and also Complements on the 9-point method)
- use of the filters generated by REW to make an amplitude correction 'automatic' by direct import into RePhase (EQ correction with REW)
- possible application of a target curve (Applying a target curve)
- creation of a minimum phase correction pulse (see above) we have all the bricks to make a very clean correction without too much ask a question

9.1.3 [Bear] measurement method incorporating the latest features of REW and RePhase integrated by their designers to make the job easier! • Download the following document: "[Bear]" method from 9 measurements

9.1.4 Comments by [Bear] on the method proposed by [Audyart] (and also taken from [Bear's] document in English above)

Correction obtained by this method, on the basis of measurements taken at the summits of a parallelepiped (8 measures). Compared with other correction systems, the results are quite impressive:
The measured phase is indeed very close to the minimum phase, and this without any retouching:

More impressive: the damping step is really much faster (the correction RePhase is in bright red).

As well as the dissipation of acoustic energy:

It is therefore an excellent method to obtain a quality correction.

9.1.5 EQ filter design in REW

The idea is to use the REW optimization engine to make life easier for us in RePhase. We start from Pascal64 amplitude measurement curve (without sub):
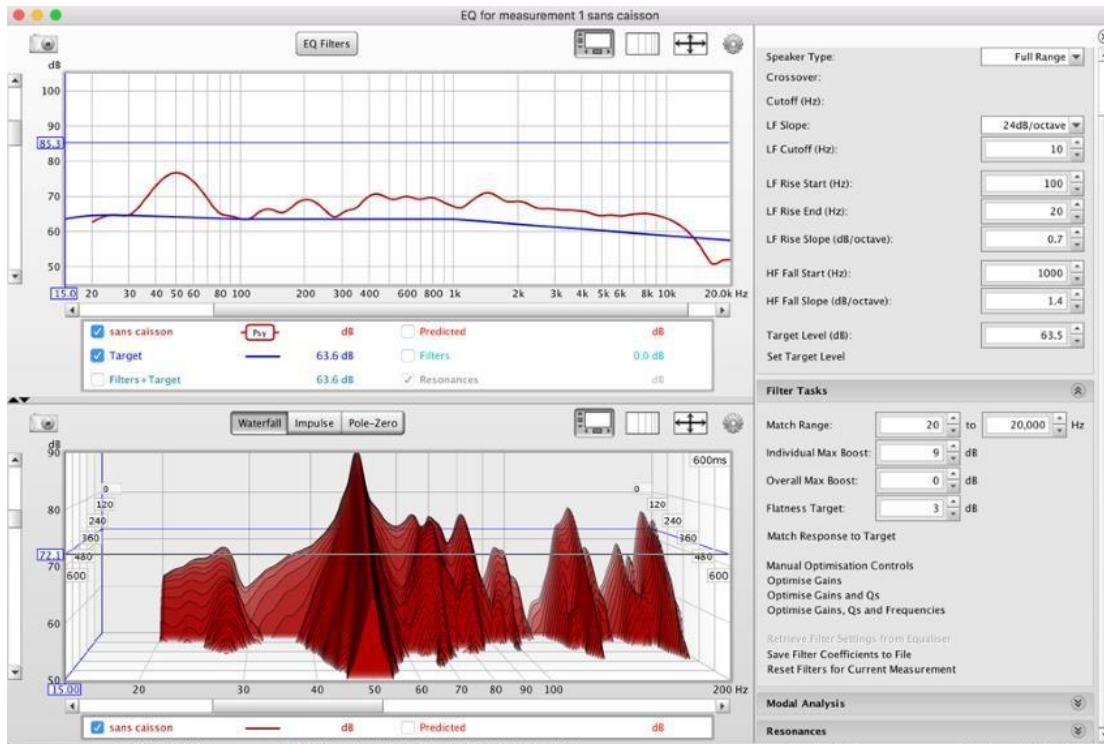

We go to the EQ tab of REW. We select 'Generic' as equalizer.

We select the desired parameters for the target curve (here a very slight slope between 20 and 100Hz, flat between 100 and 1000 Hz, and a steeper slope beyond 1000Hz, to bring a decay of about 6db between 1000 and 20 000 Hz).

We click on 'Match response to target', and magically, REW calculates the filters to apply.

Here is the corrected curve:

and the filters:

Incredibly powerful, because it only remains to copy these filters in RePhase to have
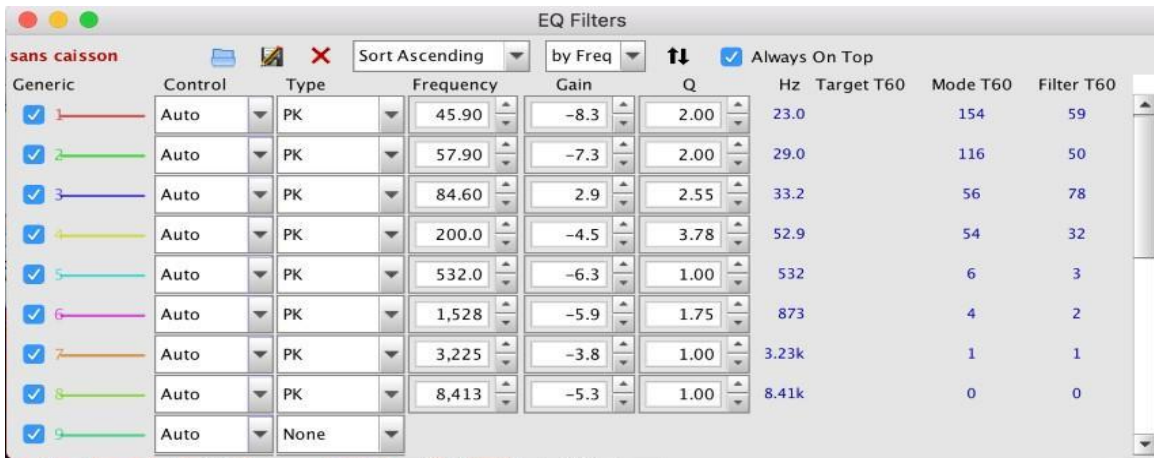a correct correction in amplitude. It remains to make the adjustment of the phase, but this tutorial has
widely spoken.

Here is a simulation of the result:
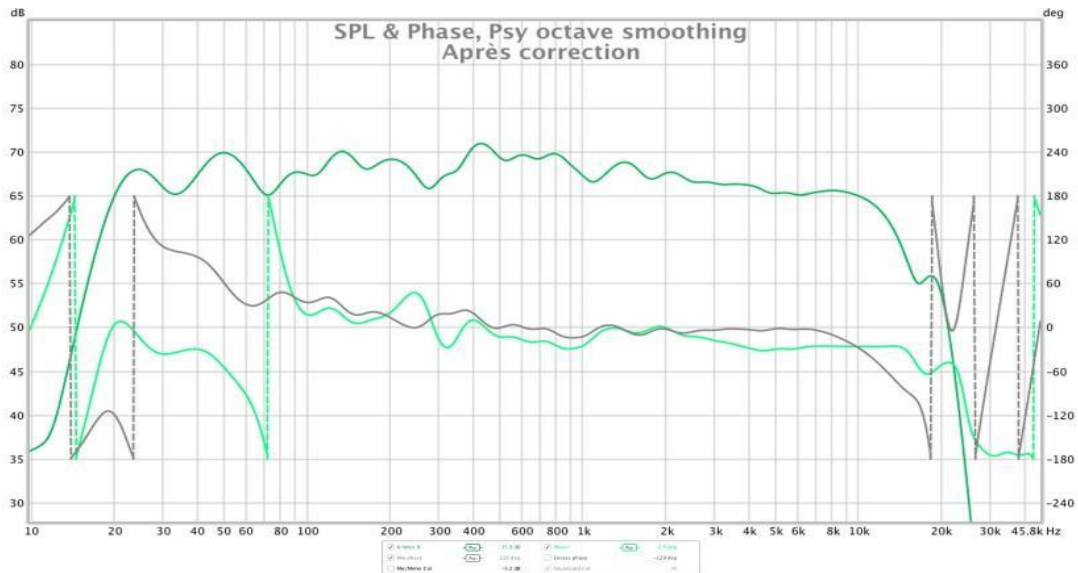
Corrected pulses:
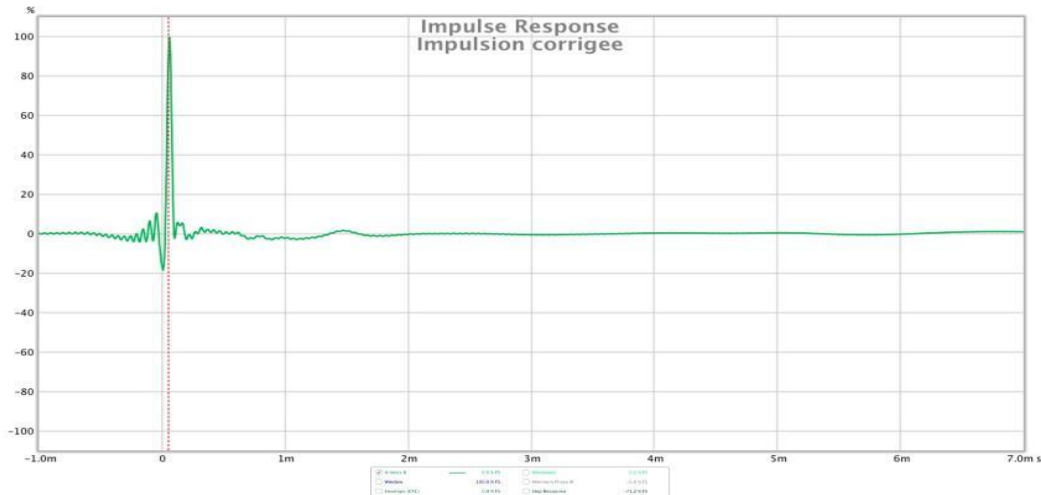
Other remarks: [Bear] Note that the amplitude correction alters the signature sound of the speakers. It is a habit to take to have a faithful reproduction. In addition, by default JRiver ticks the box 'Normalize filter volume'. Given the fact that the filters have a gain <= 0db, we can remove this check, and the volume in JRiver will be more consistent with what we are used to.
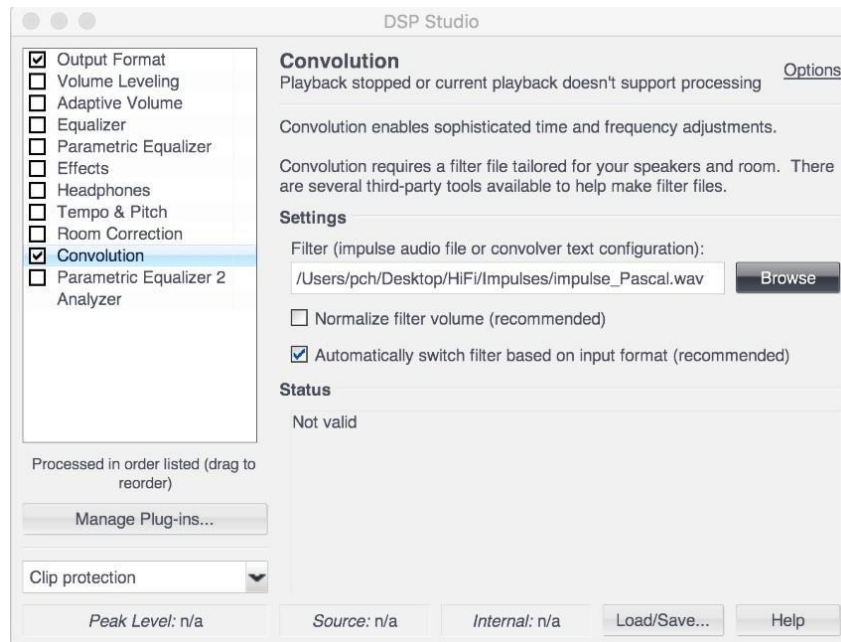
9.2 Questions / Answers
9.2.1 Q1: Single 2-way correction or one channel correction?

[Pascal64]: we triturate the neurons to make measurements and adjustments right-left for
in the end merge the two tracks in Jriver. So, only one correction or 2 corrections?

R1 [Bear]:

In a first approach, I only used your measurement of the two speakers together and generated only one mono correction for both. It is undoubtedly suboptimal, but that seemed sufficient for a first discovery. If we want to go further, free to each to make a separate correction for each channel. At home, RePhase operates with a single correction on both channels.

R2 [pda0]:

There are several possibilities with RePhase and several "philosophies". Side correction of the phase, it's normally the same thing on the 2 speakers unless, as [Bear] does, also wants to take into account some thoughts. Personally I am rather in favor correct only the speaker but the justice of the peace in this matter is listening *(Supplement [Bear] : I'm trying to replicate the minimal phase. It is no longer a question of taking into account reflections here ...)*

Amplitude side, it makes sense to independently correct each channel, but we surrender also counts with use that the correction gap is not massive, so a single correction for the 2-way walk too.

I adopted a different correction on each path (as Dirac and Trinnov do) but I think you have to watch / listen in every configuration. *(Complement [Bear]: for a symmetrical part, only one correction may be appropriate).*

Nevertheless, technically with JRiver, it seems to me that the best is the file of configuration with one pulse per channel (which can be 2 times the same if the right correction left is identical).

The advantage of leaving with the configuration files is that you can change and test easily. And as at the beginning we spend a lot of time comparing different versions it is, in my opinion, the most practical.

[Bear] sent you a single pulse in 192kHz 32bits, which is the alternative to configuration. It imposes upsampling to 192kHz but it's a single impulse to manage so quick to implement for the initial debug phase.

Personally, I do not like rendering at home and I prefer to stay in native frequency, but it must depend on the configurations at home. Moreover, it eats the power of calculation and the 192kHz with Devialet Air Wifi is not stable enough (you need RJ45 and as I prefer Wifi ...). So in summary, you have to test by ear!

9.2.2 Q2: Rephase output file type PCM for raw use FIR?

[Olivier]: can we have a PCM type Rephase output file to integrate it into a linux package that uses brute FIR for example? Does it work well with a virtual machine Windows on MacOSX, like VirtualBOX?

R1 [pda0]:
Yes. RePhase produces a ".wav" (PCM) file that is integrated in the process of "convolution". Any software that accepts this type of correction should be able to work with.

And complementary point, you only need Windows to run RePhase, but once the ".wav" is generated, you can use it on any platform (Mac, Linux, Windows, ...) which supports the reader of your choice (it is obviously necessary that this reader accepts convolution pulses in ".wav" format).

For using RePhase, I run it on Mac OSX with a virtual machine VMware in which runs Windows 7. It works without any problem and it is very convenient because everything else (REW and JRiver) are on Mac OSX at home.

9.2.3 Q3: what happens on a 24/96?
[Olivier]: what happens on a 24/96?

R1 [Bear]:
In PCM format, I did not encounter any difficulty, up to and including 65536 taps, in 192kHz. I am also aware that the convolution applied to a DSD is likely to rely on 'hidden' conversions from DSD to PCM and back, even if the designer HQPlayer does not present it this way, which is likely the cause of the limit encountered. So on the MacMini, the limit is DSD64 with 65536 taps or DSD128 with 32768 taps.

9.2.4 Q4: Convolution in Audirvana?
[pda0]: how to converse with Audirvana?

R1 [Audyart]:
After installing LAConvolver in the AU plug-in, it is recognized by Audirvana.
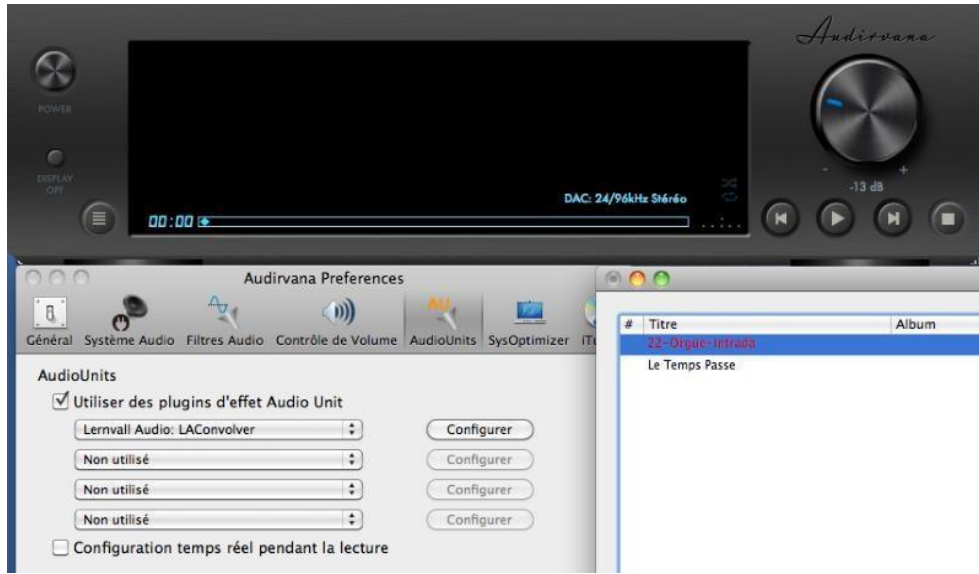
For those who would like to install it, we must put the file "LAConvolver.component" to this place: / Library / Audio / Plug-Ins / Components /

Not tried yet, but at least one weakness vs JRiver, because the convolution is only done a single frequency, so need to up-sampler if we do not want to complicate life with several configurations and a manual change ...

9.2.5 Q5: How to create active filters with RePhase?

[Pascal64]: I read that it was possible to create "active filters" with Rephase ... How does it work? Do you have tracks that I could explore?

R1 [Audyart]:

The tutorial of J. Thomas (Tuto Bass management and convective active filtering with Jriver) is a good foundation for Jriver ™:

http://jimmy.thomas.free.fr/Jriver/Tuto-Jriver-RePhase-HolmImpulse.pdf

9.2.6 Q6: alignment of HP?

[luisgomes]: I undertook to rebuild my installation to make investigations of side of the acoustic phase measurement and correct it. I acquired the UMIK-1, a sound card external USB with digital and analog I / O and measured T & S parameters of all my loudspeakers, as well as their closed response curve (roughly adapted) and 50 cm to free myself (maximum) reflections on the walls.

Having opted for the realization of 2 boxes of 190 liters to load two pairs of SPH390TC charged in push-pull closed for the bass, 2 pairs of HM130GO Audax in 4
liters closed for the medium and two silk domes of 25mm for the treble, I finalized a provision giving good results at the response curve and listening once filtering, equalization and temporal alignment applied.

I assumed that the closed charge causes less phase shift than the bass SLR and the baffle plane more reflections on the walls. All my boxes are closed. So worse for the efficiency of editing in this experiment.

In the same way, I obtained a less rugged and descending response curve more in the bass with the medium almost glued to the front wall, but I have everything even chose to place them peeled off more than a meter from this wall and side walls for minimize reflections.

I always make sure that a pulse sent simultaneously in the 3 lanes arrives rigorously at the same time, to the sample, to the measuring microphone placed at the listening. Although there seems to be controversy about the proper positioning of the pulse

of the acute (which makes 1 or 2 samples at 48k) compared to that of the serious (which can make a hundred) I consider that the impulse of the treble should be centered in the middle of that of the medium the latter two to be centered in the middle (maximum) of the impulse of the serious. I could read that the acute impulse must have been at the start of the serious impulse but if I apply an FFT pass low to a pulse, the result is a half wave centered on the original impulse.

My speakers are "globally" aligned. As they undergo a variable "delay" according to the frequency, nothing proves that they are precisely for their frequency of connection, especially in the bass-medium. At one time, I lined the speakers not not with a pulse, but with a half-wave of the connection frequency.

I wonder what method is best and about a system at the corrected phase if the two methods do not become identical?

R1 [Audyart]:

In effective IR filtering (electroacoustics) the basic logical alignment is on the beginning pulse, to match the theoretical target. In this case the emissive plans of HP are all at the same distance from the listening point. The temporal logic is causal, the excitation - cause - is at t = 0, the answer - consequence - follows.

If each channel, active, is filtered and corrected in linear phase fir, the alignment (t = 0) is on the pulse vertices, the pulses becoming symmetrical, likewise if the speaker is globally corrected to linear phase by convolution, the pathways become virtually aligned as well.

NB: Audacity makes it possible to join two mono pulses in a stereo (in 32 bt. Floating Point) for apply different left / right corrections.

[luisgomes]: Thanks for the answer, which I understand, but does not clarify my questions.

When I analyze at the point of listening a pulse restored by 2 speakers and I see in one case a hump with a peak perfectly in the middle and in another case a back donkey with a peak at the beginning, which of the 2 systems is well aligned?

My ear does not care if it's an iir filter or a fir filter that separated the signals speakers, and if the filter order is 1, 2, 4, 8 or more. I would just like to know if so that my ear hears with 2 speakers the same thing as with only one perfect wideband, the peak must be in the middle or the beginning of the donkey?

I understand that an impulse is also the sum of all frequencies (we will say 20 to 20,000 Hz to simplify and talk about audio) with increasing energy. If we filter with a filter that does not alter the phase, like an FFT in cool edit pro, we see that all these frequencies are centered on the same point, hence my tendency to want to center the peak in the middle of the donkey.

Moreover, it is obvious that a loudspeaker can start moving only after receiving the electrical signal caused by the pulse; how could he then have already have moved its membrane while the pulse has not yet occurred?

But the speaker is only an imperfect mechanism of sound reproduction, actually in a world of perfection, the membrane of the big 45cm with a mms of 300 grams should not- instantly find herself at the maximum of her swing at the moment of the impulse? But maybe just a pulse is a theoretical concept and does not correspond to no sound of real life and therefore would not be suitable for setting a restitution chain sound?

R1bis [Audyart]:

In this case (of perfection) the big 45 cm would have an unlimited bandwidth at the top, infinite for instant response.

A pulse contains all the information on the amplitude / phase response and represents this information on their temporal aspect. It's two aspects, two "angles" of view, frequency / time, (convolution / deconvolution) of the same reality.
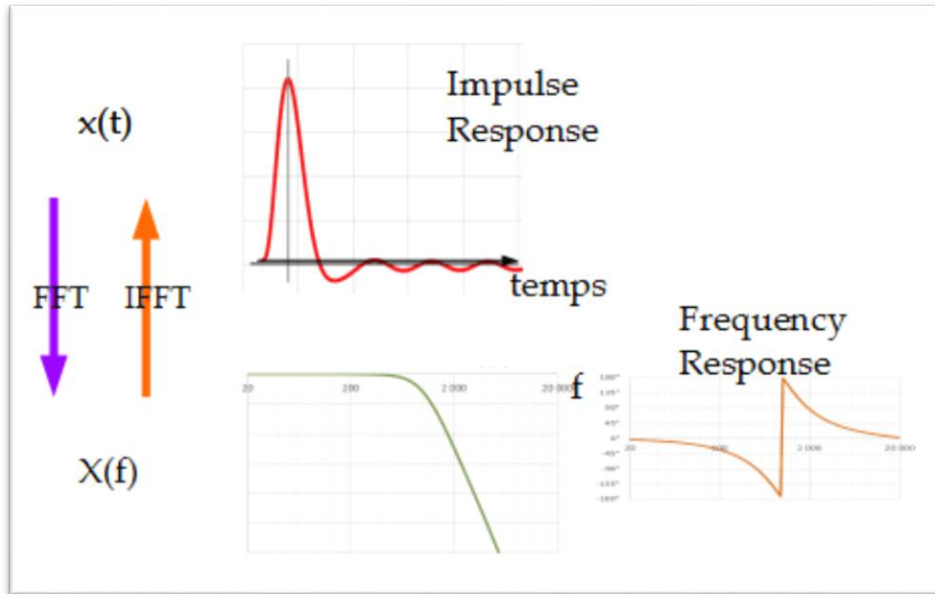
Or with the answer step - to better see the answer in the bass - which is the integral of the pulse.

9.2.7 Q7: reading the phase

[luisgomes]: I was also interested in the phase. As expected, the gross measure is unusable, the phase turns from 40000 ° between 20 and 20000 Hz. As recommended in the tutorial, I apply the proposed IR correction, the change is hardly noticeable. As proposed in the tutorial, I check "add frequency-dependent window" and choose 1/6 octave as width. The phase measurement becomes exploitable to be corrected in RePhase, but the amplitude response curve shows huge hollows that I cannot get to explain.

At this point, I ask myself several questions:

1. What is the newly calculated frequency amplitude response curve? It cannot represent the response of the speaker alone without the eliminated reflections by windowing, because without reflections, my speaker is even more linear (see 50 cm) is still that this new curve obtained by calculation cannot serve for any correction.

*2. The choice of a 1/6 octave width is arbitrary to eliminate some of the reflections. The phase curve obtained for a width of 1/8 or 1/12 will be different several hundred degrees. It's hard to admit that we can get into RePhase. **The** right phase curve to correct among the thousands of windowing possibilities for the same gross measure. I would be quite a supporter of the phase correction of the speaker alone + filters + EQ and consider that the phase distortions caused by reflections are the signature of the local listening and need not be corrected. By against, to measure, it's cotton! impossible to measure at 10 cm, the high speakers will be too far from each other for a correct measurement. It should move the exact configuration into a deaf room. At least one stage of the procedure is "feeling" excluding in my opinion any scientific rigor in this phase correction. I do not know if correcting 50% phase distortion improves listening, but if so, let's correct, it can only be better .... unless there have side effects that we do not think about yet.*

It seems that Foobar does not accept configuration files that allow separate correction for the left channel and the right channel. In view of the approximations leads to the creation of 2 separate files for the stereo, I understand better than some are content with a single file applied to both channels. The advantage of Jriver on Foobar on this point would then no longer apply.

R1 [pda0]:
Frequency dependent windowing is the only option that allows you to have a reading of the phase approximately correct by eliminating as much as possible the influence of parasitic reflections. However, it is not possible to eliminate them all. It remains all the more that you measure far from the speakers and that there are various obstacles on the way (whether on the edges, between the speakers, or possibly between the 2, like coffee table).

To do a phase correction with RePhase, as what we are looking for is just linearize the intrinsic phase to the enclosure and not the resulting one after multiple reflections (Even if windowing eliminates a lot), you can measure the speaker as close as possible to limit the impact of reflections, then window again.

In any case, you will be able to judge the "realism" of the measured phase especially as you know what you put as filters on your speaker (which is your case here) and that you can have an idea of the theoretical expectations.

An important point for the phase is to place the T0 in the right place. I place it on the peak tweeter pulse, which according to my observations, leads the phase to 0 ° to 20 kHz when we have eliminated the reflections. 1/6 seems insufficient if you have a small room and you speakers are near the walls. You can go down much lower and watch what it gives up to 1/1. On the speakers I measured at home (Giya, P3esr, Thema and Totem Arro) and elsewhere (Leedh E2), a windowing at 1/3 gives a result that seems coherent.

The resulting amplitude with windowing is, at least in theory (and according to the REW doc), amplitude as it would be without reflections, but we must not forget that some hollow observed so can also be due to the position of the microphone vs the HP, so I do not think that it is important to worry about it more than that. When window is to see the phase and correct it with RePhase.

Once you've used the windowed metric to fix the phase, just import into Reposes a measure of amplitude without windowing (File / Import Measurement) to be able to correct it, while keeping the phase corrections that you made previously.

I most often use the one that REW produces with Var Smoothing because I find that it is the one that allows a fine adjustment in the bass, and "soft" on the rest, but no matter what other option is playable!

[luisgomes]: I will continue my investigations with your last remarks in mind and maybe measure well the phase of the midrange part with the speaker almost suspended in the void in the middle of the room! I cut the sub at 140 Hz and my phase is almost flat from 12 to 120 Hz regardless of the position of the microphone. In any case, one thing that I did not integrate at the beginning, that I sensed then, and is now clarified: There is no question of correcting any modification of phase caused by the local, unlike the amplitude of which it is almost at the origin of all the faults.

R1bis [pda0]:
Yes, correct some phase errors related to local is technically possible with RePhase, but it is at his own risk on the result. I had fun doing it to correct a node linked to the room created by the interaction of the 2 speakers and the room. It works, but it produces anomalies on the amplitude elsewhere (see the result on the seats side) and I was not convinced by the result. See in particular the following link: Effects of digital correction

It therefore seems more prudent to correct only the speaker on the phase part.

9.2.8 Q8: Which format to choose when exporting Rephase?
[Pascal64]: when exporting the file to Rephase, which format to choose? In the tutorial, it says "Text format" but here we also want to export the amplitude correction.
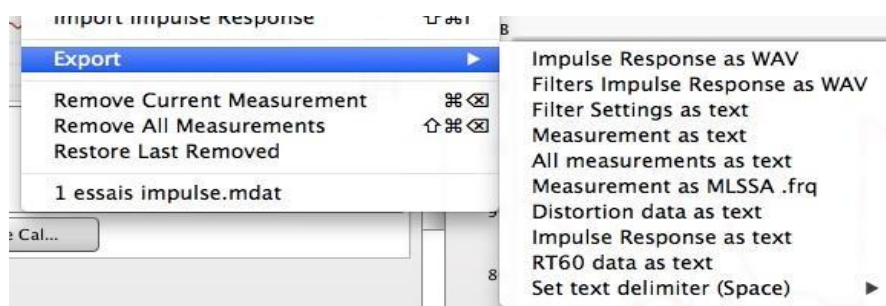

R1 [pda0]:

You have to export in txt format. REW exports the data as it is displayed. That is, according to the smoothing used and possibly the windowing. For the amplitude, it does not need to windowing (and it's the opposite for the phase) and you can use the smoothing you want. I like the Var Smoothing but other options walk too.

[Pascal64]: text format ok but
- *measurement*
- *all measurement*
- *or filter?*

I played a little rephase to try to make a special filter box Once the convolution file generated and imported into REW, here is what it gives:
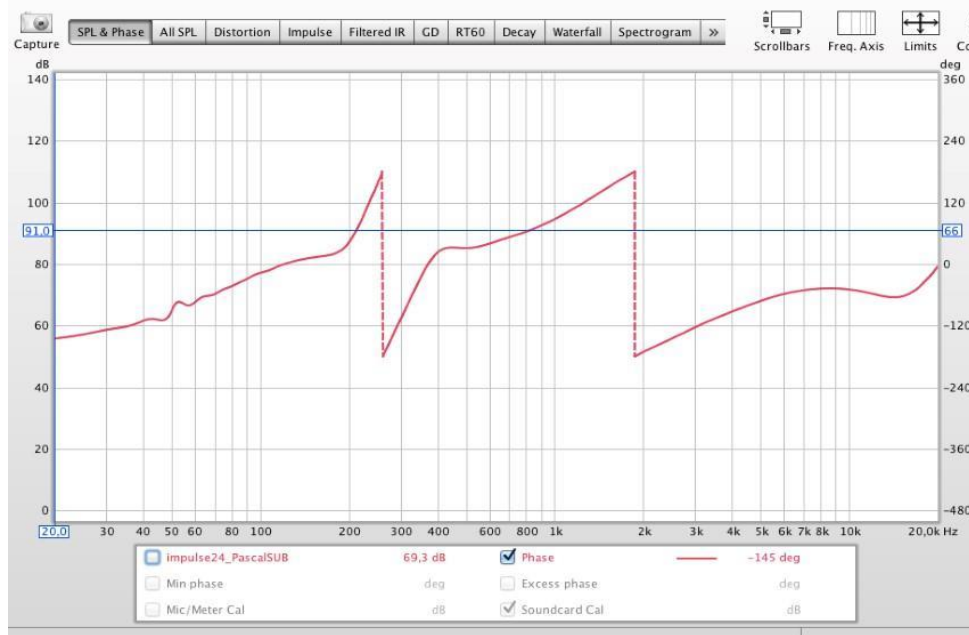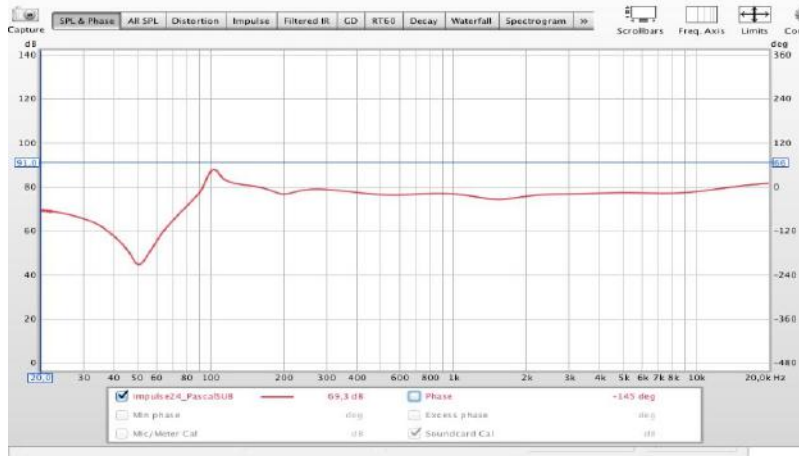
it looks like the phase has moved a little

R1bis [pda0]:
"Measurement as text" must be taken in order to record only the current measurement (the one selected). Then you give the file a name and then import it into RePhase. What you posted above is the head of the impulse which it is not easy to conclude much.

You can simulate convolution by "multiplying" the impulse with the amplitude of the measure that you used in RePhase to create the impulse, and you'll see the pace of the answer in amplitude with convolution. You have to go to ALL SPL and use the menu integrated in the display (see the tutorial on the part simulation in REW).

It's equivalent to measuring with convolution running in JRiver or Audirvana + LAConvolver, but it does not replace listening

9.2.9 Q9: amplitude / phase
[Whiterabbit]: here's what I find in rephase, I do not see how to fix this in the 'Parametric Range EQ' tab?
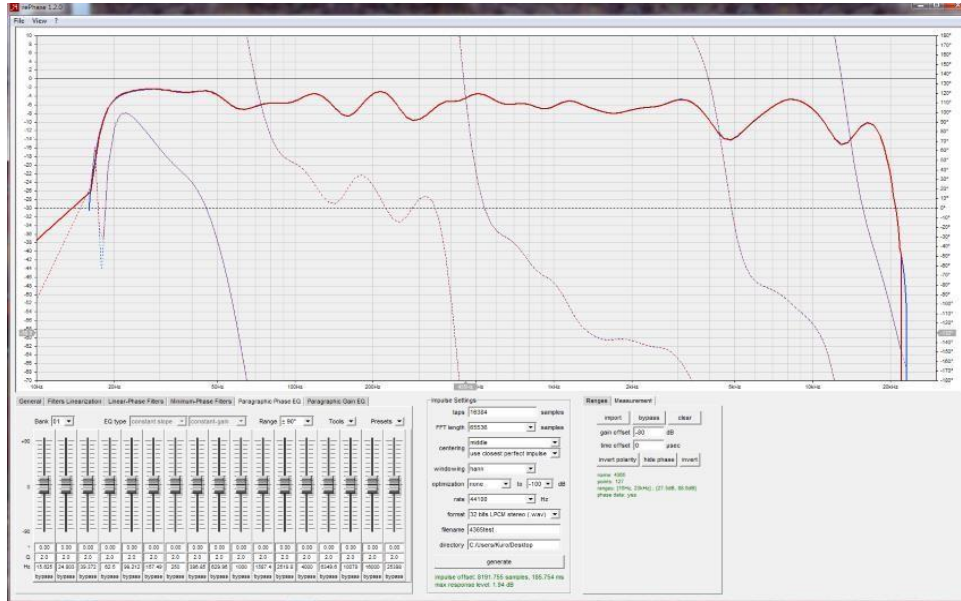
R1 [Bear]:
I do not see a Parametric Range EQ in RePhase. You have two tools, one of correction amplitude, 'Paragraphic Gain EQ', the other phase correction, 'Paragraphic Phase EQ'.

▪ If you want to correct **the amplitude**, better let REW do it with the calculation automatic proper filters. You can import the filters as described in this tutorial. ▪ If you want to correct **the phase**, before playing with the Paragraphic Phase EQ, you have to start by aligning using the 'Time Offset', the phase to zero at the Nyquist frequency (see below). Then, it will be necessary to implement the linearization of the filters before touch the equalizer.
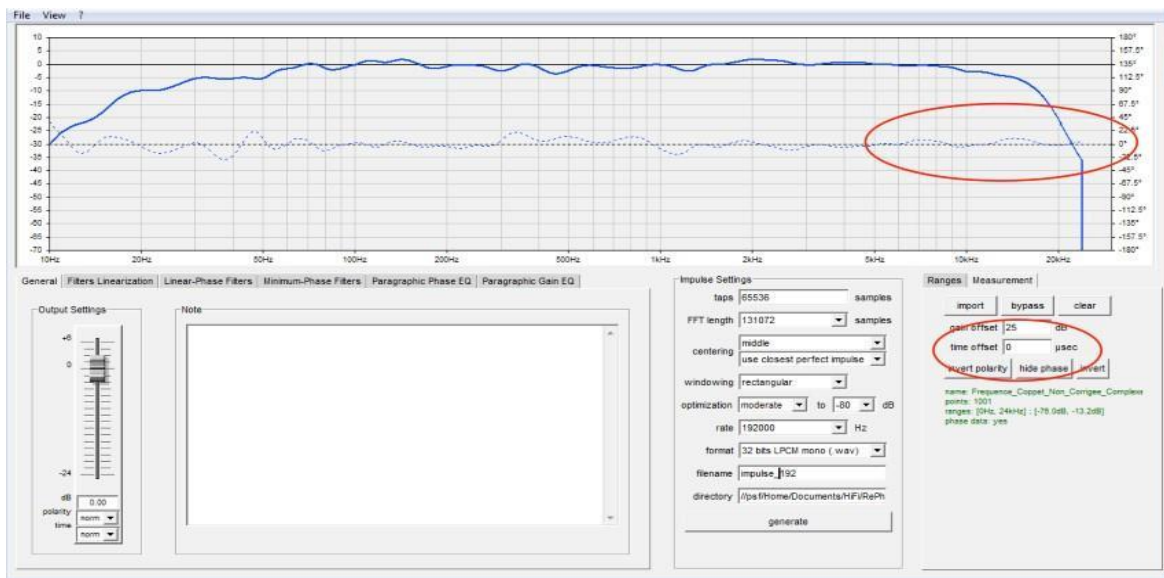
R2 [Audyart]:

In addition to the indications of [Bear], if the measurement is done with REW, you have interest in use the "estimate IR delay" function and then "shift IR" (impulse window, controls) before to export to RePhase, to pre-set the calibration of t = 0, to refine then. The rapid phase folding at 420 Hz is probably related to the conditions of measured. Does it disappear with a shorter windowing (fdw)?