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Digital equalization of automotive sound systems employing spectral smoothed FIR filters

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ABSTRACT

In this paper we investigate about the usage of spectral smoothed FIR filters for equalizing a car audio system. The target is also to build short filters that can be processed on DSP processors with limited computing power. The inversion algorithm is based on the Nelson-Kirkeby method and on independent phase and magnitude smoothing, by means of a continuous phase method as Panzer and Ferekidis showed. The filter is aimed to create a "target" frequency response, not necessarily flat, employing a little number of taps and maintaining good performances everywhere inside the car's cockpit. As shown also by listening tests, smoothness and the choice of the right frequency response increase the performances of the car audio systems.

1. INTRODUCTION

As demonstrated by the latest studies on this topic, spectral smoothing improves the robustness of an inverse filter. Automotive audio systems are characterized by very small enclosure volume and reverberation time but high Schroeder frequency comparing to rooms. In this environment, the impulse response between microphone and loudspeakers is different from a room response also because there is no separation between direct wave and reflections. This means that is not possible to equalize only the direct path but we need an inverse filter obtained considering the whole impulse response. The usage of traditional inversion techniques gives FIR filters longer or equal than the measured impulse response. Because of the limited DSP processors computing power in automotive field, we aim to reduce the filter length by spectral smoothing, as previously observed by [1].

Other advantages of this method are a remarkable enlargement of the sweet spot and the stability of the equalization. As demonstrated by [2], the performances of a smoothed filter are better than a classic filter for listening points different from the one where transfer function was measured. Furthermore they are more immune from the weak non-stationariness of the acoustic channel.

The filter synthesis is divided in three steps: spectral smoothing of the measured transfer function, decimation and inversion. The smoothing algorithm is the same of [1]. It means that there is an independent computing for magnitude and phase and this translates in a non-linear complex averaging. Each discrete quantity is averaged over a number of spectral lines using a weighting window. For the phase it is needed first to unwrap, to avoid possible errors related to discontinuities. Instead of what is shown in [1], we work in a discrete signal space, so our task is to remove discontinuities with a processor algorithm. Regarding the smoothing resolution, we followed the work of [4] and tried different resolutions. The algorithm allows for octave resolution, mixed octave-resolution, critical bands and equivalent rectangular bandwidths approaches.

Once the phase and magnitude are smoothed, they are decimated. This translates into a reduction of the spectral frequency resolution, but the action is needed to match the number of spectral lines corresponding to the inverse FIR filter that we want to get.

After the decimation, phase and magnitude are assembled to re-build a transfer function that will be inverted. The way to compute the inversion is based on [3]. The inversion doesn't change the total number of spectral points.

The same smoothing / phase unwrapping algorithm is also applied to mix many impulse responses like in [1]. The aim is to average different listening positions and so enlarge, another time, the sweet spot.

In order to obtain a comfortable equalization in automotive environment, it is required to achieve a non-flat magnitude response [5]. The proposed synthesis method takes into account this problem allowing to choose (graphically by drag and drop) a target curve for the magnitude of the equalized response.

By means of subjective listening tests, we probed a set of target curves and smoothing types.

2. FILTER SYNTHESIS

2.1. Spectral Smoothing

2.1.1. Algorithm

As previously observed by [2], spectral smoothing gives more robust results in acoustic measurements. Talking about equalization, it's easier to obtain robust inverse filters starting from smoothed measurements than from standard ones. The smoothing method is the same as [1] i.e. magnitude and (unwrapped) phase are treated in the same way.

A step-by-step explanation of the smoothing algorithm is given hereafter.

- 1 - Build a Discrete Fourier Transform (DFT) *H[k]* of a measured Impulse Response (*IR[n]*). As IR is a real sequence, it is possible to take into account only the first $N/2 + 1$ samples (if N is the length of the *IR[n]*).

$$H[k] = DFT\,(IR[n]) \tag{1}$$

- 2 - Compute the magnitude *M[k]* and the unwrapped phase *P[k]*.

$$M[k] = |H[k]| \tag{2}$$

$$P[k] = unwrap(\angle H[k]) \tag{3}$$

- 3 - Choose an averaging window shape *W[i]*.
- 4 - Choose an odd length $L = 2\alpha + 1$ for the window and compute the values of its weighting coefficients.
- 5 - Choose the spectral line to smooth *M[k]* (*k* will be the central line of a symmetric subset $M[k-\alpha]\ldots M[k+\alpha]$).
- 6 – Apply the window, and compute the averaged value *M'[k]*:

$$M'[k] = \sum_{i=0}^{L-1} M[k-\alpha+i]\cdot W[i] \tag{4}$$

- 7 - Repeat step 5 and 6 for $k = 1$ *to* $N/2+1$ and for *P[k]*.

In figure 1 there is an example of a smoothed magnitude and the original one.

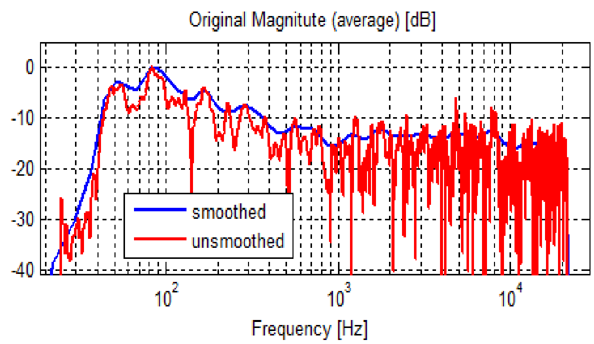

Figure 1: original and 1/3 octave-smoothed magnitude of a car frequency response.

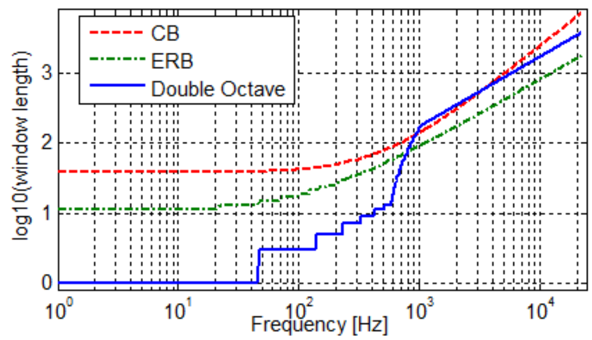

Figure 2: averaging window lengths vs. frequency. Red dotted: Critical Bands; Green point-dotted: Equivalent rectangular Bandwidth; Blue solid: Double Octave Fraction

2.1.2. Phase unwrapping

As stated previously, the phase unwrapping is a necessary step before smoothing. There are many algorithms to unwrap the phase of a DFT: Discontinuity Detecting (DD), Cepstrum based, polynomial factoring, mixed methods, etc.. To simplify the code writing we chose DD.
Phase unwrapping also allows to mix different frequency responses by simply adding up magnitudes and phases.

2.1.3. Averaging window

Of course, the averaging window can have many shapes. For synthesizing the filters we used in the subjective tests, we selected an Hanning type. More important is the discussion about the length. In non-automotive fields, some approaches have already been tested [4]. Following this example, we used some variable window lengths rules: Critical Bands (CB), Equivalent Rectangular Bandwidths (ERB), Double Octave Fraction (DOF) bands (1/24 octave below the car Schroeder frequency, ≈ 800 Hz, 1/3 octave above). The window lengths are shown in figure 2

In the double octave line, it can be observed a transition zone across 800 Hz. There are also some abrupt jumps in the other curves. It can seems worrying but remember that it's only the length of an averaging window, not a DFT magnitude.

2.2. Spectral decimation

As well known, it's possible to increase the spectral resolution of a transfer function (i.e. increase the number of points) by zero padding. Spectral smoothing allows to reduce the number of points. Once the magnitude and phase are smoothed, it's possible to decimate separately these quantities. In this way we reduce the spectral resolution and the number of points. Note that this operation is completely different from classic downsampling because it leaves untouched the sampling frequency.
Reducing the number of points is a necessary operation when you want to make a filter shorter than the IR of your system, because the inverting algorithm operates spectral line by spectral line. This does not change the number of spectral points, so you have to reduce this number before inverting.

2.3. Inversion technique

The inversion technique is based on [3]. This ensures a correct phase handling and absence of strong peaks in the filter spectrum. The inverse filter *S[k]* can be computed as follow:

$$S[k] = \frac{G^{*}[k]}{|G[k]|^{2} + \varepsilon} \cdot T[k] \qquad (5)$$

where *G[k]* is the smoothed , spectrally decimated version of *H[k]*. The regularization parameter ε has been taken typically equal to 0.01 and *T[k]* is the target curve.

A significant improvement over the original inversion method has been described in [6]. So we employ a frequency-dependent regularization parameter ε[k], and the typical spectrum of ε is shown here.

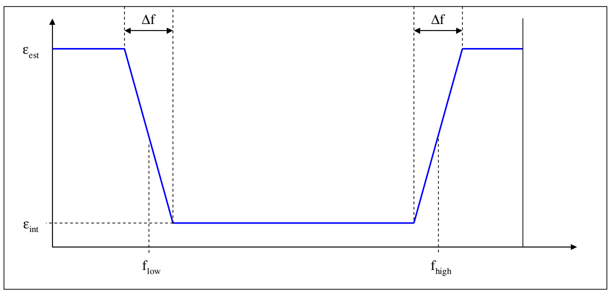

Figure 3: frequency-dependent regularization parameter

In practice, ε is left at a small value, such as 0.01, in the useful frequency range for the sound system, and is progressively increased up to a much higher value (typically 1.0) at extreme frequencies, where the system cannot be controlled, or, simply, is mute.

3. SYNTHESIS SOFTWARE

We developed a graphic Matlab[1] function suite. It allows to plot the measured frequency response and set all the filter parameters (length, spectral resolution, target curve, regularization parameters)

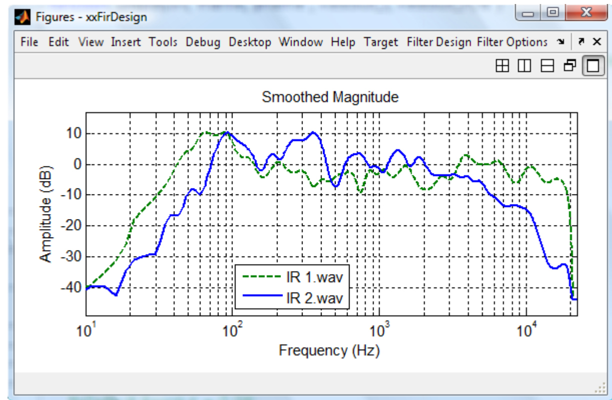

Figure 4: typical window of the synthesis tool.

As previously stated, the software allows to graphically set, load and save a target curve (see figure 4 and 5).

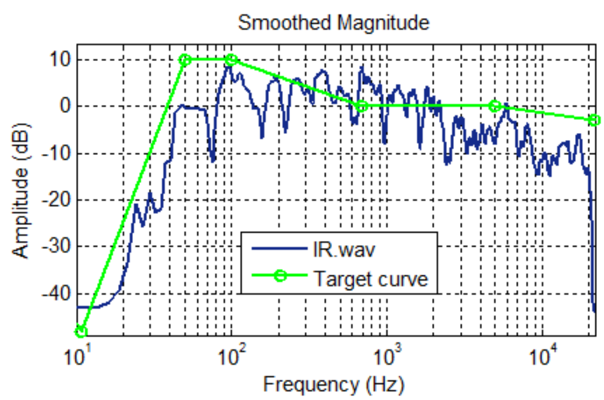

Figure 5: magnitude of a frequency response and target curve.

After the filter computation, 2 windows appear. The first (figure 6) shows 3 IRs: the original one, the inverse filter one and the convolution of these. The second window (figure 7) shows the corresponding DFT magnitudes.

[1] Matlab is a registered trademark of The MathWorks[2], Inc.

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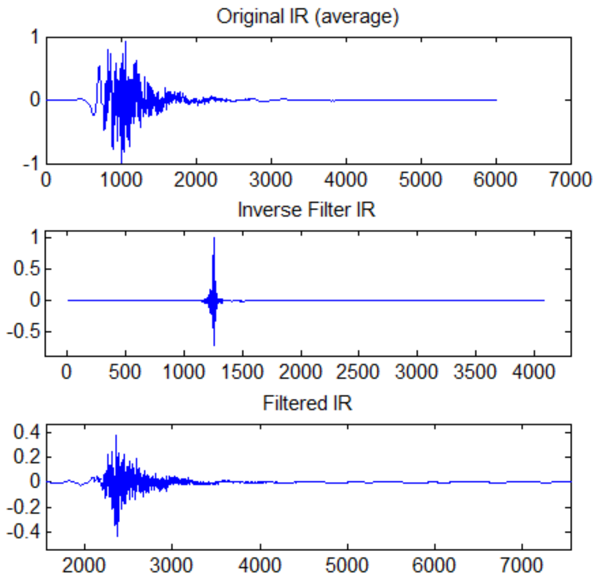

Figure 6: IRs shown after filter computation.

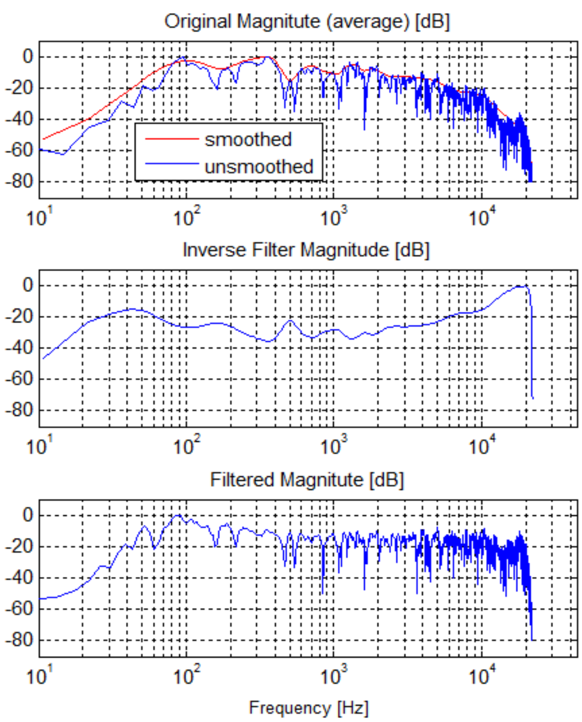

Figure 7: magnitude plots shown after filter computation.

4. SUBJECTIVE TEST

A blind listening test was performed to investigate on subject's filters liking. The 9 involved persons were medium-high skilled. In detail, we chose some target curves and averaging windows and asked the subjects to fill a questionnaire.

4.1. Chosen filters

We tested inverse filters with 2 target curves ("Soft" and "Hard")

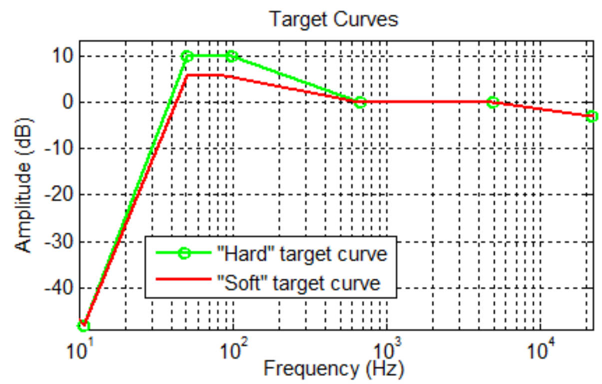

Figure 8: target curves used to build the test filters.

and 3 averaging windows (ERB, CB, DOF). Over these, the native car sound configuration (not filtered) was inserted inside the listening test. This is the test filter set:

- A – Soft + ERB;
- B – Hard + ERB;
- C – Native (not filtered);
- D – Soft + DOF;
- E – Hard + CB.

In this way we tested with at a single time both target curves and averaging windows. In figure 9 you can see some measured spectra of the transfer functions, obtained by applying a DFT over a measured impulse response of 4096 points, sampled at 44100 Hz. We payed attention to the SPL. All the sound configurations had the same global A-weighted level (within 1 dB).

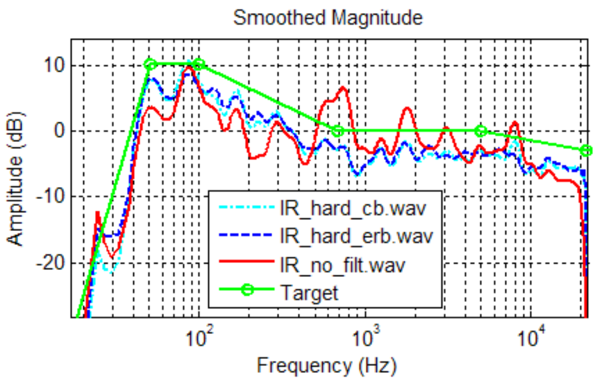

Figure 9: some measured spectra of test filters.

Digital filtering questionnaire

Filter A				
Liking		Unpleasant	o o o o o	Pleasant
Treble		Too weak	o o o o o o o o o	Too loud
Bass		Too weak	o o o o o o o o o	Too loud
Voice		Undistinct	o o o o o	Distinct
Stereo effect		Bad	o o o o o	Good
Distortion		Distorted	o o o o o	Undistorted
Warmth		Cold	o o o o o	Warm
Colouring		Coloured	o o o o o	White

Filter B				
Liking		Unpleasant	o o o o o	Pleasant
Treble		Too weak	o o o o o o o o o	Too loud
Bass		Too weak	o o o o o o o o o	Too loud
Voice		Undistinct	o o o o o	Distinct
Stereo effect		Bad	o o o o o	Good
Distortion		Distorted	o o o o o	Undistorted
Warmth		Cold	o o o o o	Warm
Colouring		Coloured	o o o o o	White

Figure 10: evaluation questionnaire (D, E, F filter omitted).

4.2. Equipment

The test was executed in a car equipped with a medium-high quality audio system. It was made by several speakers (a sub-woofer, left and right woofers, left and right tweeters and a central medium) and an automotive ASK multichannel digital amplifier. The amplifier input was connected to a stereo USB soundcard that played the audio tracks.

Thanks to an appropriate software, test subjects were able to listen an audio track and to switch in real-time the inverse filter (with no music interruption). Furthermore the software allowed to play multiple tracks. With the aim to make the test as neutral as possible, we chose four musical tracks of different genres.

The following table shows the tracks chosen:

n.	Artist	Title
1	Daugherty, Neiberg and Reynolds	I'm Confessin' (That I Love You)
2	Nirvana	Smells like teen spirit
3	Papete Disco Club	Room 5
4	A. Vivaldi	Winter

Table 1: musical tracks used in the listening test.

The evaluation questions were placed as in figure 10 (for shortness, only questions regarding A and B filters are shown).

4.3. Results

It follows an exhibition of statistic results. The first aim was to compare the filters.

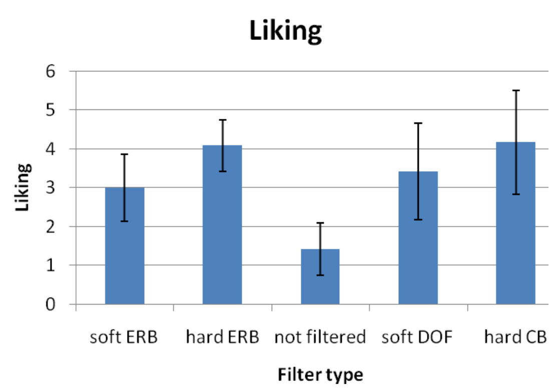

Figure 11: "filter type-liking" histogram with standard deviation indicators.

The resolution of the questions scale was 1/5 (also for bass and treble questions that were folded-up). It was hard to compare between average values results because of the big standard deviation (see figure 11). So we made a Student's t test for every couple of filters (table 2).

Filter couple	Random Percentage
A B	0,22%
B C	< 0,01%
C D	0,01%
D E	16,00%
A C	< 0,01%
A D	34,80%
A E	1,83%
B D	11,54%
B E	84,87%
C E	< 0,01%

Table 2: Student's t test on "Liking" parameter.

The random percentage is the probability that the result obtained (on "Liking" parameter) is due to casual factors. In this case the result is not significant.
In table 2, the rows evidenced have a percentage lower than 5%, indicating that the perceived differences between these filters are significant.

Other results come from the research of a linear relationship between subjective parameters, in detail between the global parameter "liking" and the others. You can see the relationships yielding correlation coefficients greater than 0.8 in figures 12 to 16.

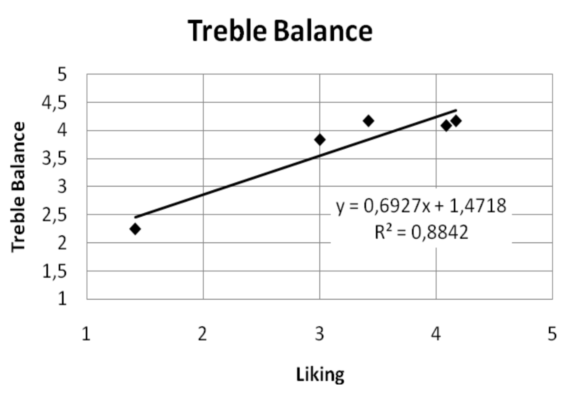

Figure 12: "liking-treble balance" relationship.

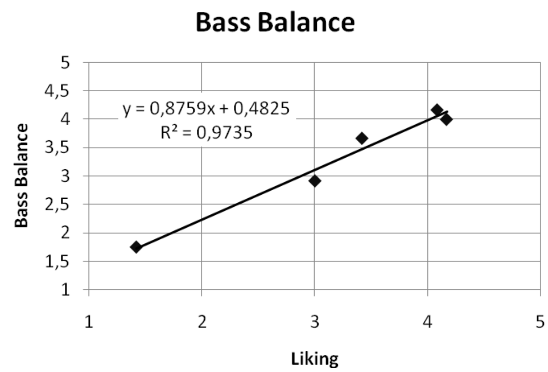

Figure 13: "liking-bass balance" relationship.

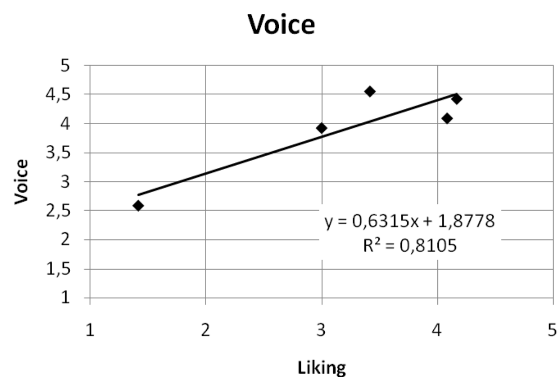

Figure 14: "liking-voice" relationship.

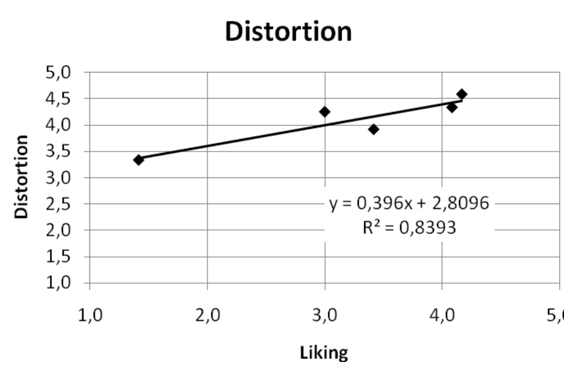

Figure 15: "liking-distortion" relationship.

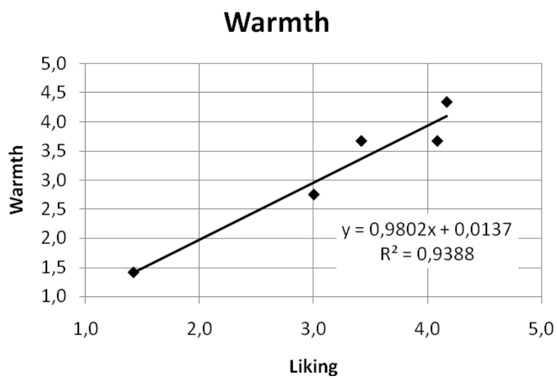

Figure 16: "liking-warmth" relationship.

We also researched a relationship between subjective parameters and objective ones. The objective parameters we used are the octave levels of the measured impulse response. You can see the relationships yielding correlation coefficients greater than 0.7 in figures 17 and 18.

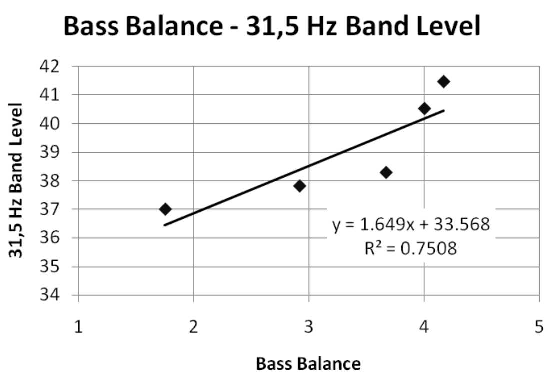

Figure 17: "bass balance-31,5 Hz band level" relationship.

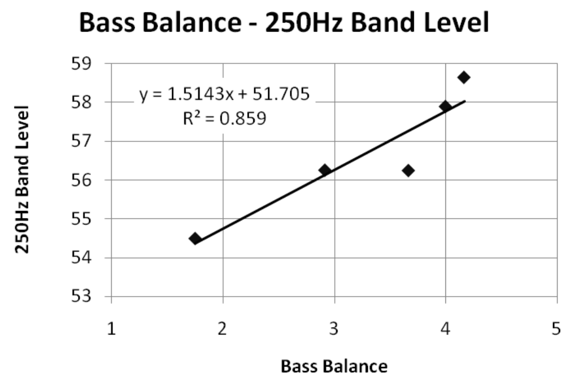

Figure 18: "bass balance-250 Hz band level" relationship.

Using as objective parameters the spectral roughness and the spectral distance from the "hard" target curve, we obtained the results of figure 19 and 20.

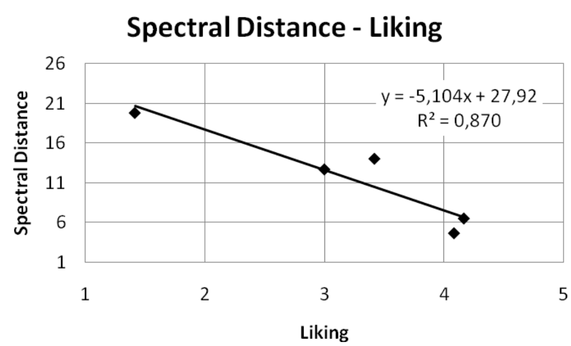

Figure 19: "liking-spectral distance" relationship.

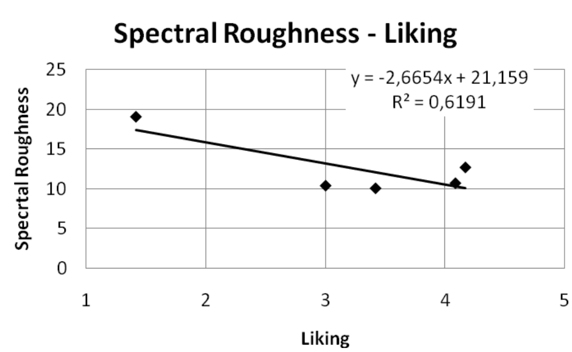

Figure 20: "liking-spectral roughness" relationship.

5. CONCLUSIONS

A comparison between filters with different magnitude shapes and spectral smoothness types has been done inside a car.
From Student's t test, we can state that all digital filters that have been used clearly improve the subjective liking of the played sound.
We can also say that the filters with "hard" target curve are the best between the tested configurations. You can still see this from the Student's t test (raw A B) and from the relationship between "spectral distance" and "liking" (figure 19).
Unfortunately, we cannot establish if there is an averaging window better than another but only that filters liking increases with the smoothness of the spectrum (figure 20). Further investigations will be done on smoothness types.
Other interesting results come from subjective parameters relationships. We found 5 adjective well related to the global filters liking.
Also good relationships between the subjective parameter "bass balance" and octave level bands have been found.

6. ACKNOWLEDGEMENTS

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7. REFERENCES

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