Audio Engineering Society
Convention Paper
Presented at the 112th Convention
2002 May 10–13 Munich, Germany

Different approaches for the equalization of automotive sound systems

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ABSTRACT

The paper discusses the implementation of various possible DSP algorithms on low-cost platforms suitable for mass series production of automotive sound systems. The analysis takes into account traditional IIR and FIR filtering schemes, dual-rate and hybrid approaches, and new algorithms such as Warped FIR (WFIR) or frequency-domain partitioned convolution (BruteFIR). The comparison is critically made checking the implementation, cost and performance of different processing schemes, but also addressing the hot problem of computing the optimal filtering coefficients for each of these schemes starting from measurements taken inside the car cockpit, making use of examples taken from the real life. The results show that both IIR and FIR structures are capable of good results on low-cost DSP systems; the more advanced algorithms will probably become competitive as soon as a new generation of floating-point DSP processors will also be available for low-cost applications.

INTRODUCTION

One of the major application of Digital Signal Processors (DSPs) in audio systems is the acoustic equalization. This is not limited to the flattening of the overall frequency response, but it is a more extensive concept. In certain case the goal of the equalization is to provide a prescribed frequency response curve (which in general can be very different from the flat one). In other cases, also the time-domain response is important, and the goal of the equalization can be described as a re-alignment in time domain (or the linearization of the phase response, which are equivalent options).

Special issues are to be addressed when the equalization is to be applied to a multi-channel sound system, and when there are many listeners in different positions.

All these issues are related to car sound systems also, where further specific hot topics need to be addressed:

- The optimal frequency response requires a substantial boost of the low frequencies, in order to ensure a reasonable S/N ratio in presence of loud background noise, which has approximately a “brown” spectrum.
- The frequency response should be adjusted according to the vehicle speed and the engine RPM, since the background noise is strongly variable and depends on these variables.
- The listeners are located off-axis with respect to the loudspeakers. Moreover the loudspeakers are often placed in “strange” positions. This results in a blurred stereo image and may cause perceived coloration because of the (Head Related Transfer Function) HRTF filtering of wavefronts coming from not-symmetrical directions.
- The available electrical power is limited. Therefore a too heavy equalization causes an unacceptable drop in signal amplitude, making the reproduced sound too weak, and causing distortion if

the user is forced to raise too much the gain control of the amplifier, in order to compensate the loss of dynamic range.
- Dealing with an entry-level sound system, the car manufacturer fixes strong constraints on the total cost of the sound system, and also restrains the size, weight and power consumption.

In summary it is impossible to achieve at the same time all these requirements: some trade-offs are needed. This paper discusses in detail the design of DSP-based equalizers to be included in automotive sound systems, based on practical experience done employing both fixed-point and floating-point processors, and making use of different algorithms for performing the equalization.

DSP ALGORITHMS FOR AUTOMOTIVE SOUND SYSTEM EQUALIZATION

In this paper, the following DSP algorithms are taken into account and compared:

- *IIR structures* (typically in the form of bi-quad 2nd order filters). These recursive filters are functionally equivalent to their analog counterparts, and they allow the implementation of general frequency response by cascading different sections, each being configured as a typical filter type (LP, HP, BP, BR), and specifying for each section the cut-off frequency f_c, the sharpness factor Q, and the gain G (the latter being relevant only for fixed-point implementation, because with floating-point processing a single overall gain can be applied at once, either before or after the cascade of IIR sections).
- *FIR structures*. These structures are also defined convolution filters, and are usually implemented as repetitive sum and multiplication, by means of a tightly optimized MAC (Multiply-Accumulate Cycle), which is usually taking just one DSP clock cycle for each coefficient on modern architectures. The length of the impulse response (usually called the number of taps) is in the order of some hundredths.
- *Dual-rate FIR structures*. These filters are obtained employing a short FIR for equalizing the higher frequencies, while simultaneously the signal is down-sampled and processed with another FIR filter for the low frequencies. Finally the two processed signals are merged back together. This topology addresses specifically the fact that a short FIR filter is not capable of an effective equalization at low frequencies, while the drawback is that the overall structure is more complex and that some errors appear in the cross-over region [1].
- *Warped FIR*. These structures closely reproduce a standard FIR filter, except that the delay unit is replaced by an all-pass section. By proper choice of the warping coefficient λ incorporated in the all-pass sections, the frequency axis is distorted, giving more resolution to the low frequency range, and compressing the high frequency range in a little number of spectral lines. This allows an accurate low-frequency equalization, at the expense of a coarser equalization at high frequencies. Moreover the computational cost of each cell of the computing network increases [2].
- *Frequency-domain convolution* (Select-Save). This approach is based on the use of large FFT blocks, which allows the implementation of long filters, thus overcoming the constraints imposed by processor size. This is usually done with the Select-And-Save algorithm, which requires the use of FFT blocks being long at least twice the length of the impulse response (typically several thousands of samples). This results also in a large processing latency, so that the filtered signals is substantially delayed with respect to the input signal. Another drawback of this algorithm is that it requires very large memory blocks, for storing the input and output buffers, the intermediate data and the coefficients [3].
- *Partitioned frequency-domain convolution* (BruteFIR). This algorithm is quite old, since it was first applied in 1966, when the limited memory available was preventing from using the theoretically more efficient Select-Save algorithm. It was re-implemented only very recently, because it was discovered that, on modern processors, this turns out to be more efficient than the Select-Save, giving the additional advantages of required much less memory, and reduced input-output latency [4].

In summary four of the above six structures are conceptually similar, being always different implementation of FIR convolution. The others instead (IIR and WFIR) are recursive structures, which require more care for designing the filtering coefficients so that the required transfer function is created, without causing instability.
In the following chapters an applicative example of the performances of the first two structures is given. All the examples are referred to the equalization of the sound system of the same car, and consequently the results can be compared directly.

DSP PLATFORMS

ASK Industries developed two independent multichannel power amplifiers for automotive applications, incorporating a DSP-controlled filtering section. The first one is a 4-channels amplifier equipped with a Texas Instruments TMS320C54 processor (16 bits, fixed point, 100 MIPS). This unit is shown in the following figure 1.

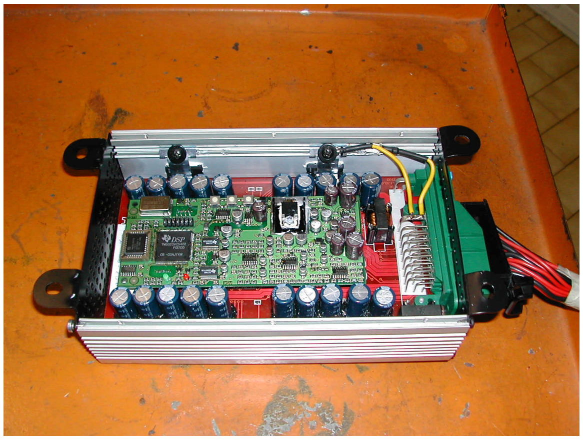

Figure 1 ASK 4-channels integrated DSP amplifier (DIGIcar).

The second one is a 6-channels amplifier, with digital input and 6-channels output, equipped with an Analog Devices SHARC 21065L processor (32 bits, floating point**,** 60 MHz, 180 MFLOPS). This second system was not employed in this work, but in the next future it will be evaluated in strict comparison with the first one.

SOFTWARE TOOLS

An automatic procedure for the synthesis of filter coefficients was developed. It relies on state-of-the art algorithms of signal processing and is based on a friendly user interface, which allows the user to check the effectiveness of its equalization before the actual measurements.
This tool, named DIGItools is a stand-alone Win32 application, which allows the generation of 4-channels 7-band IIR filters, and of 4-channels FIR filters. As far as IIR filters are concerned, for each of the 7 bands the tool allow to specify the center frequency, the filter Q, and the filter gain. Each of the above mentioned parameters can be modified by simple sliders (fig. 2).

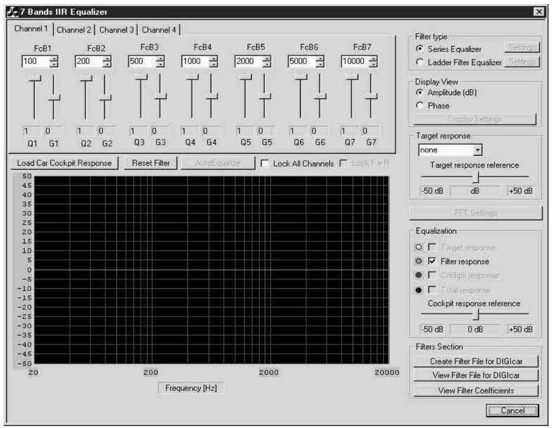

Figure 2 DIGItools IIR design screenshot.

As an example in the following some screenshots from DIGItools are presented. They describe step by step the definition of a parametric equalizer, starting from car acoustics measurements. In the screenshots the measurements of the car SPL (Sound Pressure Level) is reported together with the frequency response of the parametric equalizer and the estimation of the car frequency response with the current equalization. Figures 3-6 report the above-described screenshots for FL, FR, RL, RR channels of a FIAT Stilo, used as a test car.

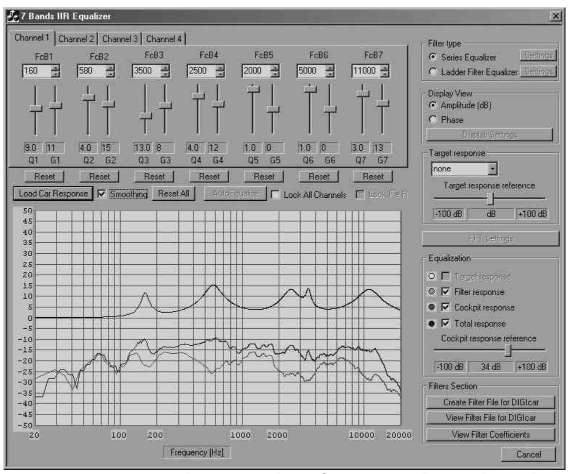

Figure 3 DIGItools IIR equalizer for FIAT Stilo FL channel.

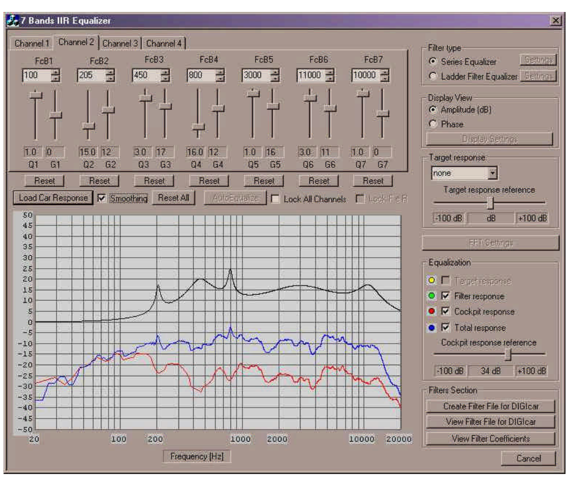

Figure 4 DIGItools IIR equalizer for FIAT Stilo FR channel.

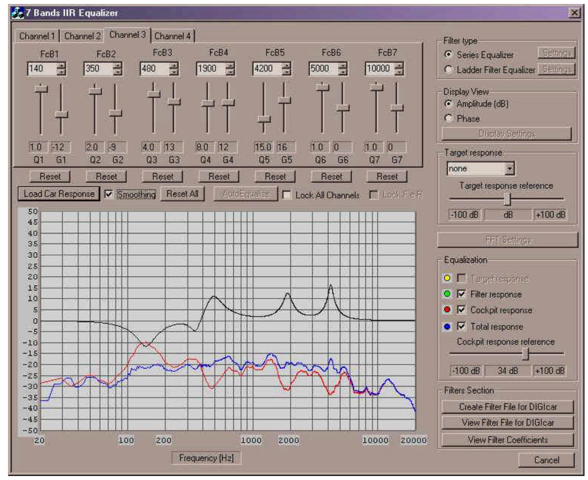

Figure 5 DIGItools IIR equalizer for FIAT Stilo RL channel.

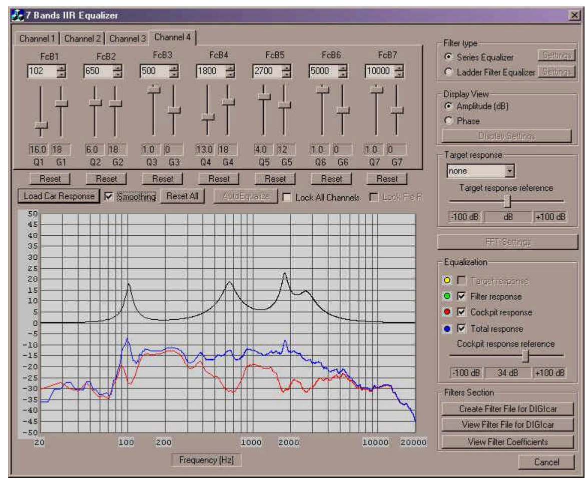

Figure 6 DIGItools IIR equalizer for FIAT Stilo RR channel.

Figure 7 shows the FIR equalizer tool. It allows to load the car measured SPL and to choose among a few inversion algorithms, like Kirkeby [5], Neely-Allen [6], Mourjoupolos [7].

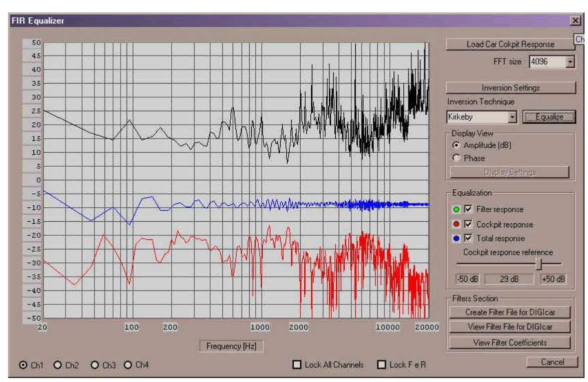

Figure 6 DIGItools FIR equalizer for FIAT Stilo FL channel.

EXPERIMENTAL RESULTS

Experiments were performed on a test car to validate different equalization methods with the 1st dedicated hardware designed by ASK (DIGIcar, shown in fig. 1). The test car was a FIAT Stilo, equipped with a standard audio system. The DIGIcar DSP amplifier was inserted in the system driving directly the loudspeakers and each of the four channels was operated separately.

Standard acoustics measurements were performed with the aid of the Aurora suite of plugins for CoolEdit [8], and they were processed by DIGItools to obtain suitable inverse filters, and operated to realize a both parametric IIR equalization and Neely & Allen FIR equalization.

The SPL measurements (before equalization) obtained are shown in the following figures. Fig. 8 shows the impulse response of the FL channel. Figs. 9-12 show the SPLs of the four channels in the frequency domain (with a resolution of 1/3 of octave).

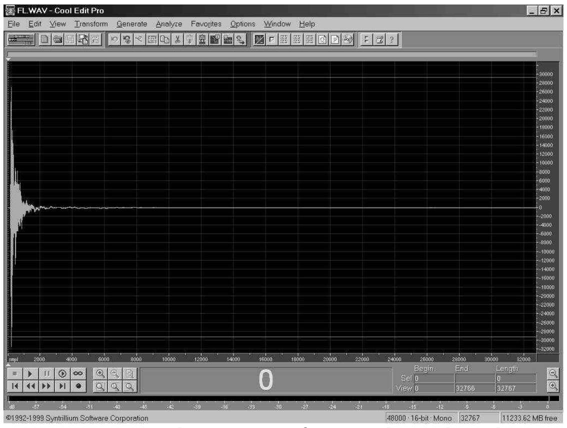

Figure 8 Impulse Response of FIAT Stilo FL channel.

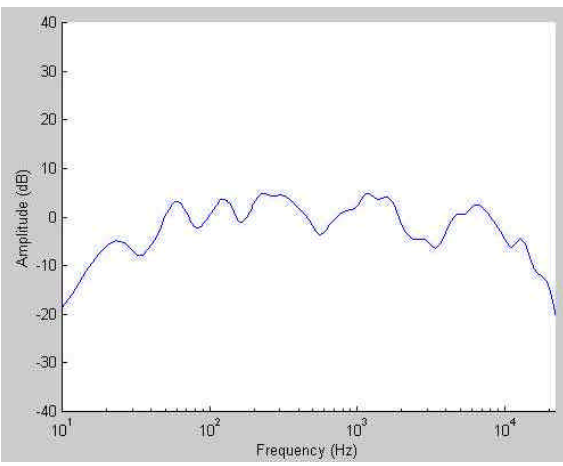

Figure 9 Frequency Response of FIAT Stilo FL channel

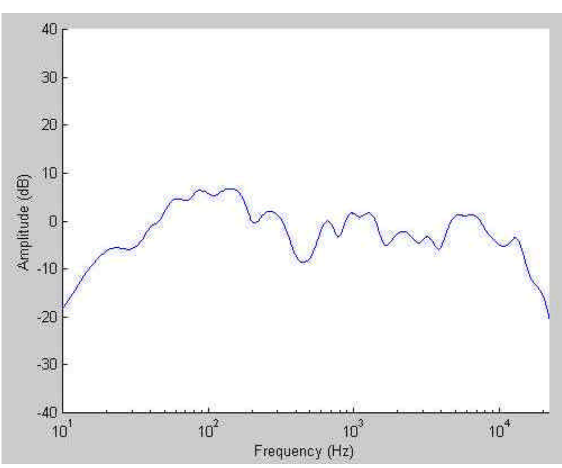

Figure Frequency Response of FIAT Stilo FR channel

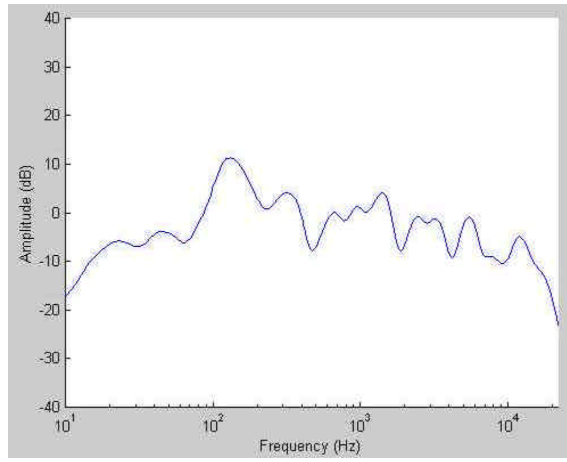

Figure 11 Frequency Response of FIAT Stilo RL channel

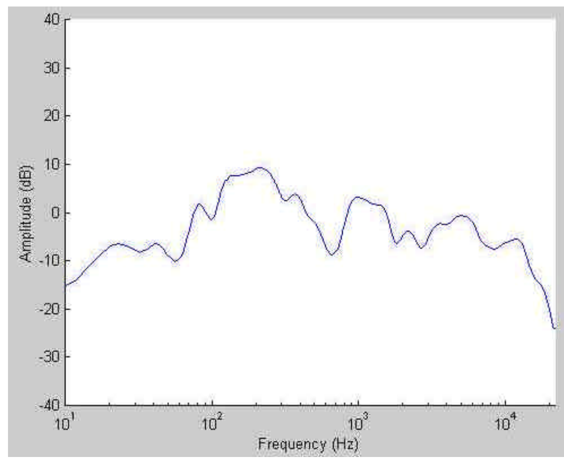

Figure 12 Frequency Response of FIAT Stilo RR channel

The inverse filters computed by means of the DIGItools software, starting from the measured car SPLs, were implemented on the DIGIcar platform by means of an assembly coded, highly optimized program. Each of the 4 FIR filters was 387 taps long, and each of the 4 IIR filter was made of 5 2nd order sections. Figs. 13-16 show the electrical measurements of the DIGIcar unit programmed to realize the four inverse FIR filters. An Audio Precision System 2022 was used for measuring these 4 electrical transfer functions.

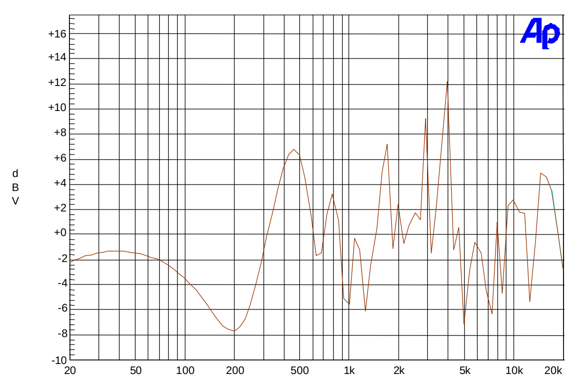

Figure 13 Electric measurements of DIGIcar FL channel programmed to synthesize the FIAT Stilo inverse FIR.

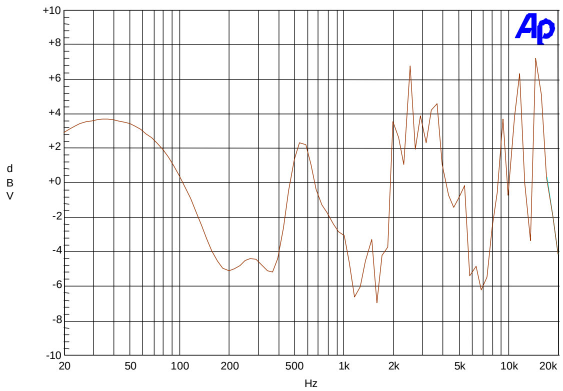

Figure 14 Electric measurements of DIGIcar FR channel programmed to synthesize the FIAT Stilo inverse FIR.

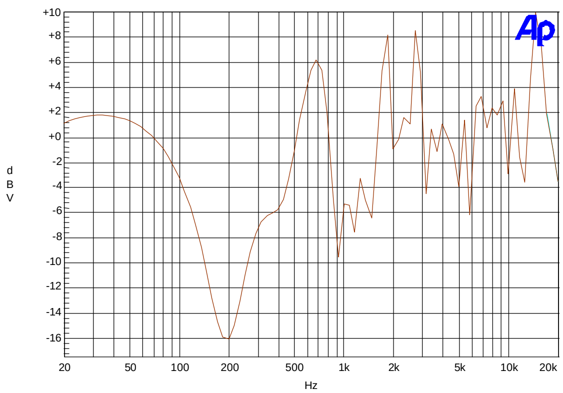

Figure 15 Electric measurements of DIGIcar RL channel programmed to synthesize the FIAT Stilo inverse FIR.

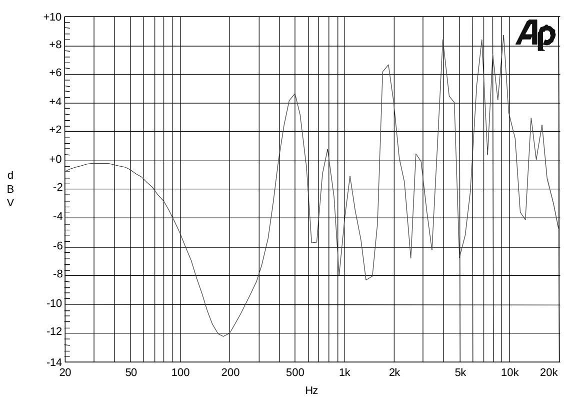

Figure 16 Electric measurements of DIGIcar RR channel programmed to synthesize the FIAT Stilo inverse FIR.

Figs. 17-20 show the results of the SPL measurements made inside the FIAT Stilo while the DIGIcar equalizer was operating. The target curve was set as the flat one, and a noticeable flattening of the spectrum can be seen, especially at medium and high frequency. Each figure reports single channel measurements with and without the digital equalization.

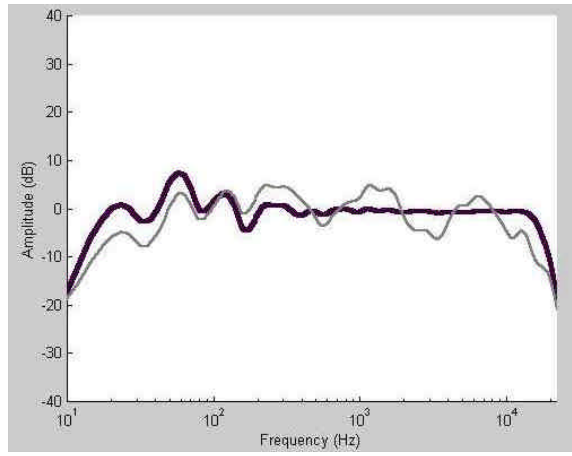

Figure 17 Frequency Response of FIAT Stilo FL channel with (thick line) and without DIGIcar FIR equalization.

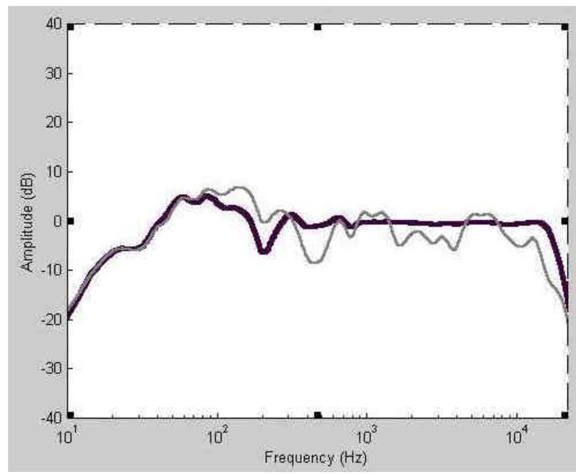

Figure 18 Frequency Response of FIAT Stilo FR channel with (thin line) and without DIGIcar equalization.

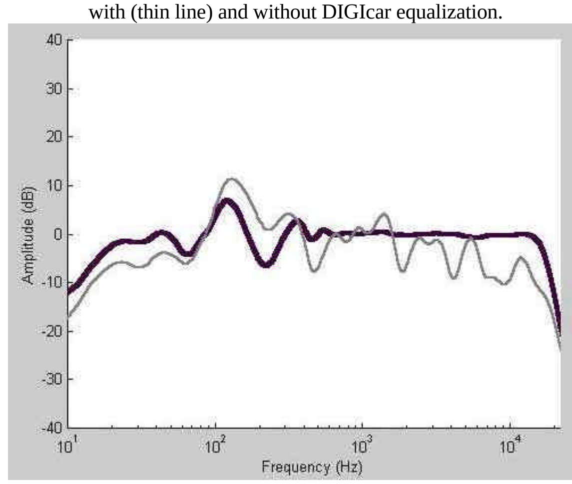

Figure 19 Frequency Response of FIAT Stilo RL channel with (thick line) and without DIGIcar equalization.

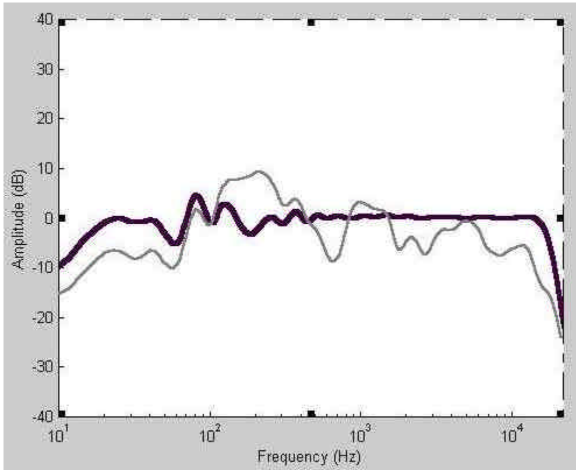

Figure 20 Frequency Response of FIAT Stilo RR channel with (thick line) and without DIGIcar equalization.

As far as the IIR equalization is concerned a set of parametric equalizers was designed with DIGItools. It resulted in 5 biquad IIR filters for each channel, which were implemented on the DIGIcar platform. Figure 21 show the electric measurements of the four channel programmed to realize the parametric equalizers.

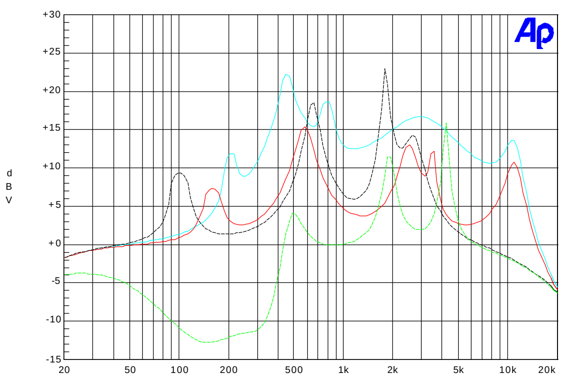

Figure 21 Electric measurements of DIGIcar channels programmed to synthesize the FIAT Stilo IIR parametric equalizer.

Subjective results were performed also. Listening tests were performed within the car (Fiat Stilo) equipped with DIGIcar equalization. The IPA procedure was used [9]. Five questions are made to the listeners, and for each of them the answer range from a minimum to a maximum:

1) Spatial Sound, Flat Sound;
2) Insufficient Level, Sensible Level;
3) Clear Treble, Unclear Treble;
4) Weak bass, Clean bass;
5) Pleasant Sound System, Unpleasant Sound System.

13 listeners were used for the IPA experiments.
The final results were IPA = 7.2 with FIR equalization and IPA = 6.9 with IIR equalization (fig. 22).
The same car will also be available during the 112th AES Conference in Munich for performing additional listening tests, which will make it possible to enlarge the number of subjects.

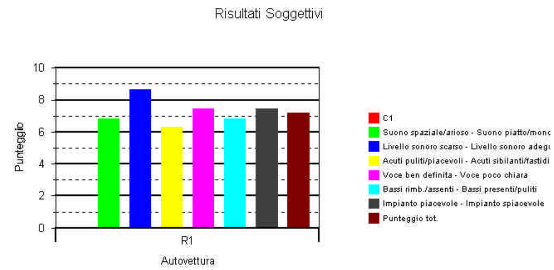

Figure 22 IPA Subjective results for the IIR equalization.

CONCLUSIONS

An automatic procedure for the computation of the coefficients to be employed in both IIR and FIR digital equalization for car sound systems is presented. It allows to compare the two different structures from both implementation and algorithmic point of views.
Experiments were performed on a commercial car to show how it is possible to obtain very quickly a nice automatic equalization and to test it with both subjective and objective tests.
The example presented here refers to an equalization aimed to flattening the frequency response, but the developed system allows also to equalize for a different target curve, as it is common to do regarding the optimal equalization of a car sound system.
The research will therefore prosecute, with the goal to assess the performance improvement obtainable by the use of a DSP-controlled power amplifier under normal usage conditions (i.e., with the background noise of the car running on the road), and to select the equalization algorithm which will give the better results at the listening tests.

References

[1] G. Cibelli, A. Bellini, E. Ugolotti, A. Farina, C. Morandi, “Experimental validation of loudspeaker equalization inside car cockpits”, 106th AES Convention, Munich, 8-11 may 1999.
[2] A. Bellini, G. Cibelli, E. Armelloni, E. Ugolotti, A. Farina, “Car cockpit equalization by warping filters”, IEEE Transactions on Consumer Electronics, vol. 47, no. 1, pp. 108-116, February 2001
[3] Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989
[4] A. Torger, “Brute-FIR – an open-source general-purpose audio convolver”, http://www.ludd.luth.se/~torger/brutefir.html.
[5] O.Kirkeby, P.A. Nelson, P. Rubak, A. Farina, “Design of Cross-talk Cancellation Networks by using Fast Deconvolution”, preprint of 106th AES Convention, Munich, 8-11 may 1999
[6] S.T. Neely, J.B. Allen, “Invertibility of a room impulse response”, J.A.S.A., vol. 66, pp. 165-169 (1979).
[7] J.N. Mourjopolos, “Digital Equalization of Room Acoustics”, Journal of AES, vol. 42, n. 11, November 1994, pp. 884-900.
[8] A. Farina, F. Righini, “Software implementation of a MLS analyzer, with tools for convolution, auralization and inverse filtering”, 103rd AES Convention, New York, September 1997.
[9] E. Ugolotti, G. Gobbi, A. Farina, “IPA - A Subjective Assessment Method of Sound Quality of Car Sound Systems”, Proceedings of the 110th AES Convention, 2001 May 12-15 Amsterdam, The Netherlands.