Car cockpit equalization by warping filters

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Abstract

Sound reproduction within a car inside is a difficult task. Reverberation, reflection, echo, noise and vibration are some of the issues to account for. A first step in the direction of increasing sound comfort is that of equalizing the acoustic pressure response in the frequency domain. A direct solution is that of implementing with a convolution filter the inverse of the acoustic measurements in the car cockpit. However this often results in very high number of taps, therefore more complex techniques must be adopted, which rely also on psychoacoustics considerations. In this paper an automatic tools to develop warping inverse filters for target car cockpits was designed and validated through experiments in commercial cars.

Introduction

A first step in the direction of increasing sound comfort is that of equalizing the acoustic pressure response in the frequency domain. To accomplish this task the inversion of the measured Sound Pressure Level **(**SPL**)** should be performed [1], [2].
Therefore two main step are involved. The former is composed by the acoustic characterization of the target car cockpit, in terms of amplitude and phase of the acoustic pressure. The latter is based on the synthesis of the filter that accurately reproduces the inverse car frequency response with a finite number of taps, and on the implementation of the audio processor on a DSP platform.

From direct measurements of typical car impulse response it can be induced that a nice inversion of the SPL requires a large number of filter coefficients. This is of course prevented from the low-cost hardware structure available. Therefore an option is that of reducing the computational cost of a simple digital filter, in order to solve the computational issue, keeping safe the inversion capability and obtain a fair equalization, which can be implemented on a low-cost DSP [3]. This can be achieved exploiting fast convolution techniques for discrete-time signals, such as overlap and save, overlap and add techniques, which operate in the frequency domain and gain from the use of Fast Fourier Transformation.
Another option is that of exploiting signal processing techniques to realize a better filter implementation. To this aim warping theory can be exploited.
In this paper the latter approach was investigated, and a prototype digital board which implements warped filters (WFIR) was designed. Several simulations and experiments were performed in commercial cars to evaluate the effectiveness of the equalization by WFIR, and compare the results with FIR or Multi-rate FIR filter equalization. The prototype designed is being tested on commercial cars for the next generation car sound system.

1 Frequency Warping theory

The basic idea of the warping theory is well represented in the following chart. In fig. 1 the FIR-

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like structure (WFIR) is reported [4], while on fig. 2 is reported the structure adopted for our purposes.

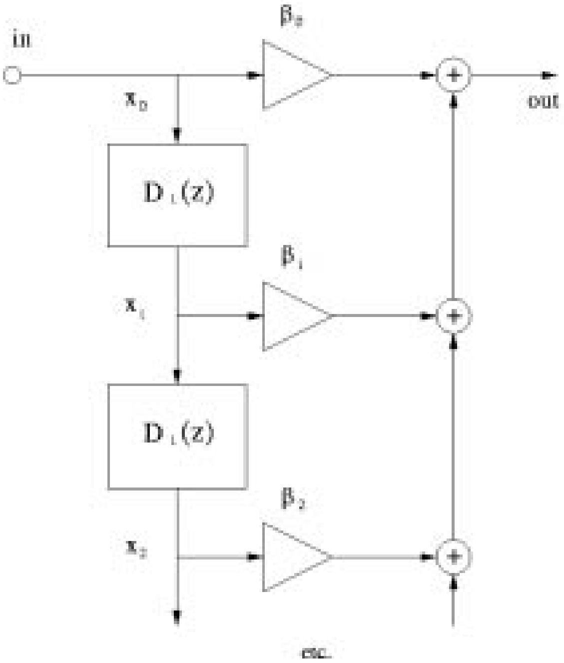


Figure 1 WFIR filter structure general structure.

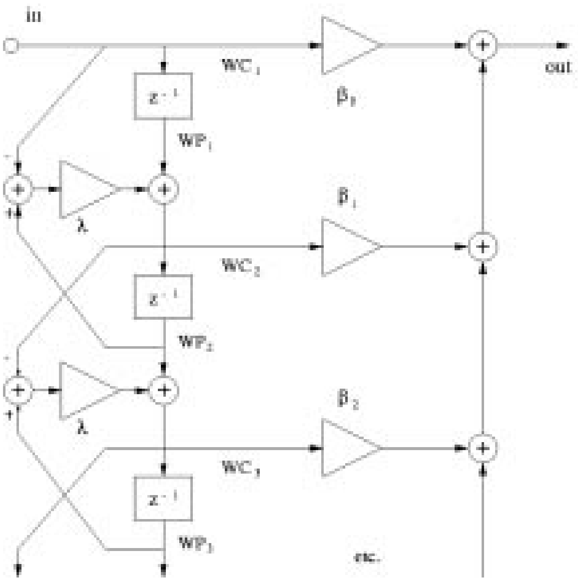


Figure 2 WFIR filter structure basic blocks.

A generic WFIR is introduced from a standard FIR filter when the delay units in a standard FIR filter are replaced by all-pass first-order functions $D_1(z)$. The WFIR is fully characterized by the
bilinear transform

$$D_1(z) = \frac{z^{-1} - \lambda}{1 - \lambda\, z^{-1}}$$

which allows to transform the frequency plane into the “warped“ frequencies. λ is referred to as the “warping parameter“ and it ranges from -1 to +1, and $D_1(z)$ is the warped delay unit. Since it is frequency dependent, the frequencies are remapped so that keeping the sampling rate constant the resolution increases for the low part of the spectrum, with a positive value of λ, while negative values increase frequency resolution for high frequencies. This transformation is consistent with the logarithmic resolution of the human hearing system.
Fig. 3 shows warping effect produced by a first-order all-pass structure as a function of frequency.

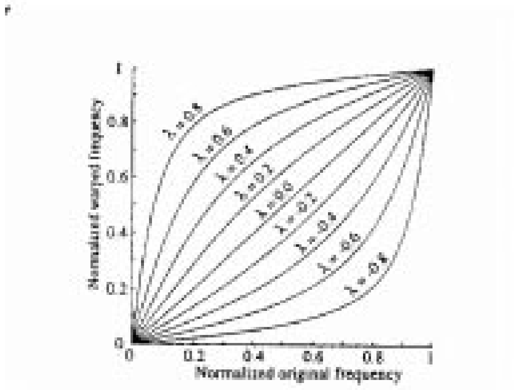


Figure 3 Bilinear transformation of frequency with different λ values

The choice of λ allows to tune the warping level, and from an acoustic point of view, it allows an approximation of the Bark scale [5]. The best value of λ depends on the sampling frequency, and Smith [6] provides to this aim an analytical expression. For instance when the sampling frequency f_s is 44.1 kHz λ is around 0.8.
Advantages and drawbacks of warped filters are addressed by its specific differences versus “classical“ filters. In a classical filter the frequency resolution is constant in all the frequency range, fig. 4. Since human hearing sensitivity is so that frequency resolution is of about one third of octave the equalization effectiveness is nice at higher frequency and bad at lower. This rule of thumb means that when sampling is uniform the low frequency part of the signal needs to be tailored more accurately than the high frequency part.

Therefore the application of warping transformation to the frequency range with a positive value of λ results in a transformation of the frequency spectrum more fitted with human hearing sensitivity. A warped filter on the Bark scale allows a better resolution and a more effective equalization at low frequencies.

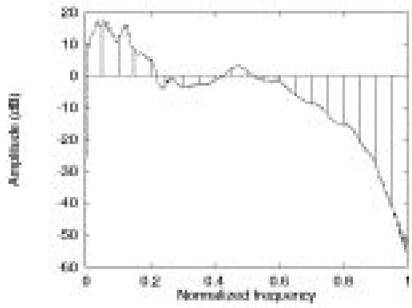


Figure 4 SPL shape of a target environment.

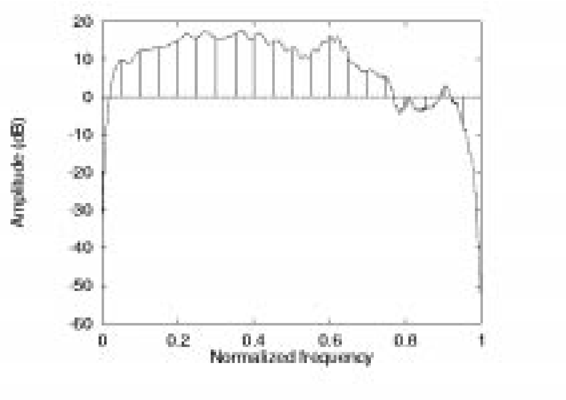


Figure 5 Pre-warped SPL **(**$\lambda < 0$).

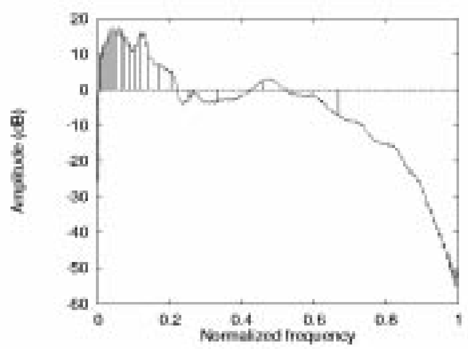


Figure 6 SPL synthesized with a WFIR.

As an example fig. 4 shows the SPL of a car inside, which can be processed by warping theory. This can be made relying on the structure reported in fig. 2 and obtaining a new spectrum (fig. 5). At this step the shape can be operated by the same WFIR structure as before but with opposite λ, and the shape reported in fig. 6 is obtained. This last chart clearly states the increase of resolution in the low part of the frequency spectrum.

The WFIR can be directly designed relying on a logarithmic psychoacoustics scale, like Bark scale [6] and this results in a better approximation of human hearing resolution. Another advantage is that usually the WFIR order is lower than the "classical" counterpart, once fixed some performance benchmarks. Nevertheless as stated by fig. 3 the complexity of a WFIR is larger than a FIR, and this results in an increase of computational cost. Specifically the computation of a WFIR requires about five or ten times more the a FIR, depending on the DSP platform adopted.

The best trade-off is dependent on the application, WFIR is especially fitted for low-frequency equalization with a rather low number of coefficients.

In the time domain the transformation of the frequency domain results in a compression of the information related to the low frequency part of the spectrum in the first samples. This holds especially for high values of λ and it means that a low number of coefficient of the step response can be kept, without affecting the accuracy of reproduction of the low frequency contents.

This is shown in the following through some simulations. Fig. 7 reports the impulse response of a loudspeaker in an anechoic chamber.

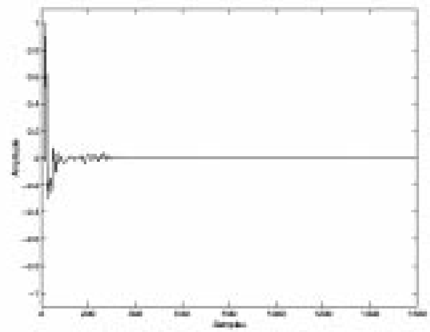


Figure 7 Loudspeaker impulse response, measured in an anechoic chamber.

It also represents the taps of the FIR filter which can approximate the loudspeaker behavior. If be abruptly truncate this sequence, for instance keeping only the first 30 samples, the amplitude of the system frequency response is strongly distorted, as graphically reported in fig. 8.

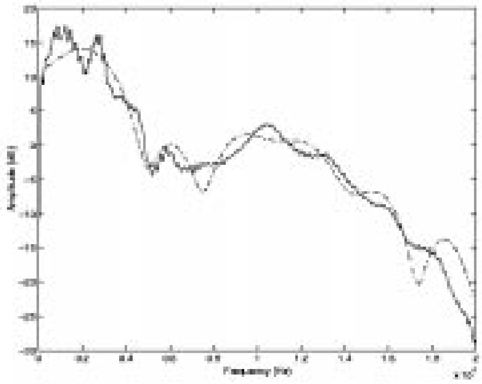


Figure 8 Amplitude of the loudspeaker frequency response before (solid line) and after the truncation (dashed line).

Fig. 9 on other side shows the impulse response obtained with the WFIR structure, with $\lambda = 0.8$. As expected from theory basis, some sort of compression arises. Information is packed in the first samples, while the whole duration increases.

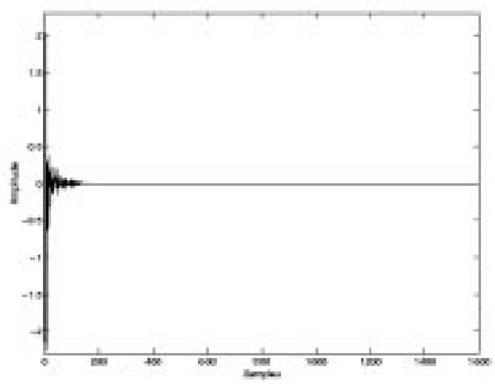


Figure 9 Warped Loudspeaker impulse response, measured in an anechoic chamber

If the same abrupt truncation after 30 samples is applied to the warped signal, the spectrum is kept almost unchanged especially in the low frequency part, fig. 10. The high frequency part, on the other side, is completely different from the original one.

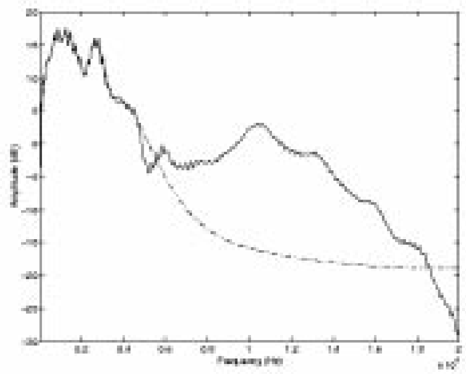


Figure 10 Amplitude of the warped loudspeaker frequency response before (solid line) and after the truncation (dashed line).

2 Car cockpit equalization

The aim of equalization is to obtain a flat band frequency response, therefore a system transfer function $H_{inv}(f)$ should be added to the audio source, so that the overall response is linear. This requires the assumption of a linear system, which is not the case due to loudspeaker instrinsic structure. The basic car audio system is depicted in fig. 11, and in the following it will be considered a two-channel linear system.

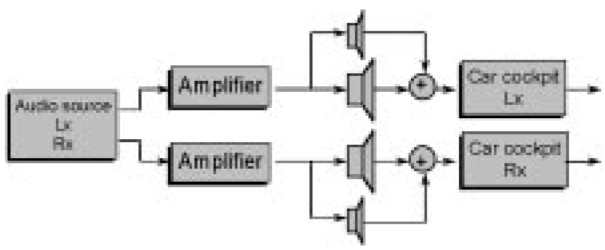


Figure 11 Car audio structure

This means that the equalization should be applied only to the part of the signal spectrum where non-linear distortion is not significant.

The adoption of a filter which realizes directly $H_{inv}(f)$ is not the best solution, and can lead to unexpected drawbacks. Moreover it could be even dangerous to use an equalizing filter which employs the $H_{inv}(f)$ directly.

In fact psychoacoustic considerations on human earing system suggest the following items

- An ideal flat band reproduction system is not necessarily pleasant. In general an increased low frequency response is desirable, especially as far as car-audio systems are concerned, where running car noises must be masked.
- Human hearing system resolution is logarithmic and limited at about one third of octave.

From these it stems that the target frequency response function should be not exactly flat, but consistent with some sort of tailored equalization response. This should be confirmed by listening test and by expert considerations, see fig. 12.

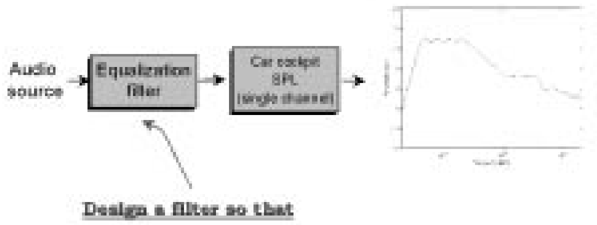


Figure 12 Target filter structure.

From human hearing resolution it stems that is uneffective to equalize all the frequency range with the same resolution. In fact at low frequencies the human hearing system can tell between different sounds which are effectively near in frequency, while at high frequencies even the most sensitive ear fails.
Therefore the equalization of very narrow peaks or holes in the frequency response would be inefficient and even unpleasant.
Therefore to reproduce human hearing resolution the inverse harmonic response $H_{inv}(f)$ a median filter was applied [7] with a frequency window slightly shorter than a third of octave.
These operations result in a new frequency response with reduced complexity, while the perceptual contents is almost unchanged if compared with the measurements data, fig. 13.
In order to achieve the target frequency response defined above, the $H_{inv}(f)$ processed by the median filter is multiplied by the target frequency response.
Moreover an ideal equalizing system should have a dynamic range of about 30-40 dB. This means that the insertion of such a device before the amplifier could overcome the maximum allowable dynamics of the others devices, with a high possibility of arising distortion phenomena. To avoid this drawbacks it is necessary to limit the operating range (say for exemple ±6 dB depending on the reprodiction system dinamic), so that the correction would be pleasant, smooth and natural.

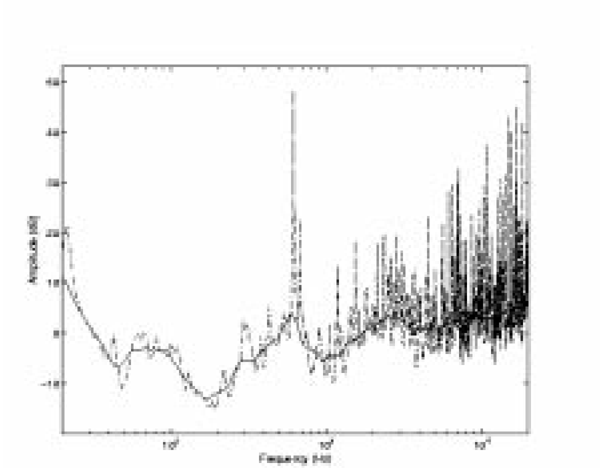


Figure 13 Amplitude of $H_{inv}(f)$ (dashed), and its processed version (solid).

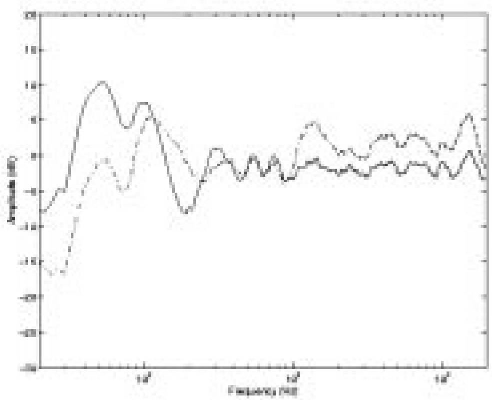
3 Experimental results

The warping theory was exploited to realize a modified FIR structure. The structure, reported in fig. 2 was implemented in DSP-based platform.
Then a semi-automatic procedure composed by the following steps was used to generate the equalizer coefficients.

1. Measurements of the impulse response on the target loudspeaker.
2. Computation of the “optimal” value of λ, which is a function of the sampling frequency, [7]
3. Transformation of the impulse response in the “warped” domain. This can be made using the WFIR structure reported in fig. 2. This requires the β_i coefficients of the original impulse response, and the application of $-\lambda$. The impulse response of this filter, which may be referred to as ”pre-warping” is the impulse response of the warped filter to be implemented in real-time. The same result can be obtained applying the transformation reported in the previous section to the signal.
4. Synthesis of the taps of the ”warped” filter.
5. Synthesis of the car cockpit inverse filter.

These steps were embedded in an automatic development tool, which produces the WFIR

coefficients for each target cockpits, acoustically characterized with standard measurements (step 1).
Therefore the final structure of the warped filter, which implements SPL^{-1} is the same as fig. 2, where β_i are the taps obtained with the above mentioned procedure and λ is the optimal warping coefficient (with positive sign).
A Citroen Evasion and an Alfa 156 were used as target environment. The acoustic measurements performed within the car cockpit by means of Aurora [8] are processed at first with Matlab, in order to verify the effectiveness of different warped filters.
Simulations show that warping filters can be successfully used for car cockpit equalization, and that the quality of the frequency response flattening increases with the order of the filter.
It can also be noted that a very low number of taps is sufficient to obtain nice results.
All experiments were performed at 44.100 kHz.
As for the Evasion cockpit the experiments were performed adopting the WFIR equalizer described and specifically 27 taps were used. Nevertheless it must be considered that the FIR equivalent to a 27 WFIR requires about 135 taps in terms of computation complexity, with the chosen hardware platform.
In figs. 14 the frequency response within the car for both the channels are shown with and without the digital FIR equalization.
In figs. 15 the frequency response within the car for both the channels are shown with and without the digital WFIR equalization.
It stems that the equalization accuracy especially at low frequency is much better with the WFIR filter, while the computation complexity is similar.



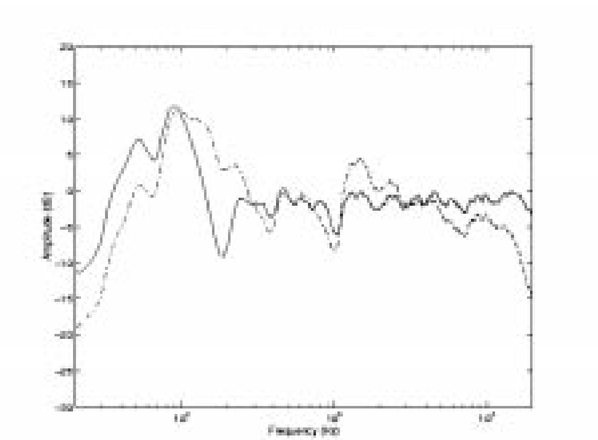


Figure 14 Citroen Evasion SPL with 350 FIR equalization (solid line) and without (dashed line), left channel (top) right channel (bottom) .

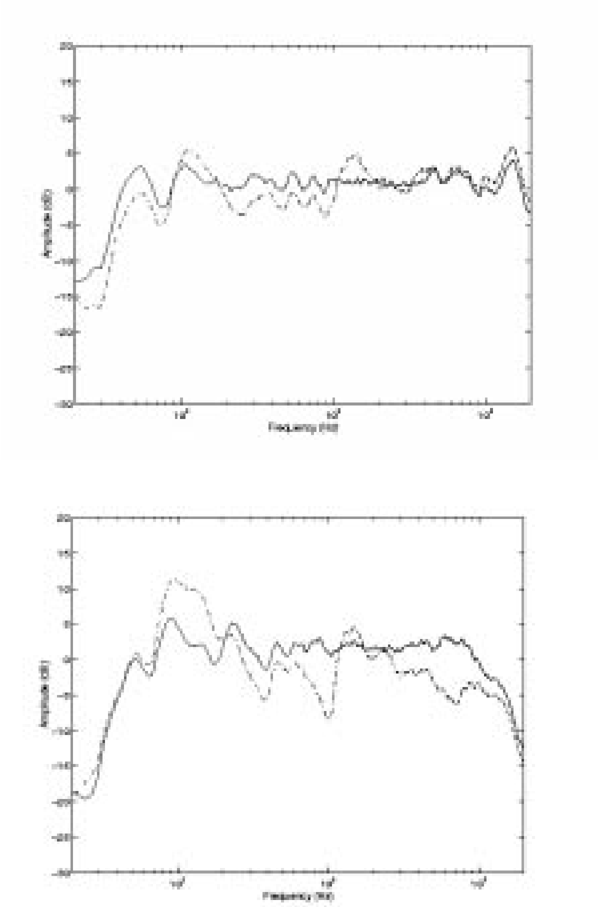


Figure 15 Citroen Evasion SPL with 27 WFIR equalization (solid line) and without (dashed line), left channel (top) right channel (bottom).

The experiments were performed adopting the hardware and software structure described in the previous sections. Specifically a 27 WFIR filter was used, and in fig. 16 the used test structure is sketched.

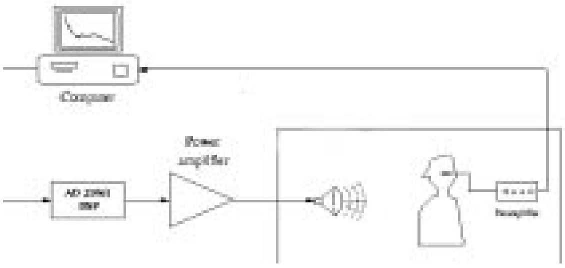


Figure 16 Experimental test set-up.

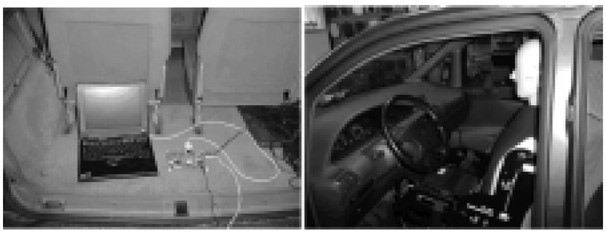


Figure 17 Citroen Evasion cockpit. Dummy head in the driving position (right) and Digital audio processor (left).

Another experiment set-up was made for the Alfa 156 cockpit, in order to compare three different filter structures fixing the computational cost. The choices were a 89 WFIR (corresponding to about 460 clock cycles, λ=0.77), a 451 FIR (corresponding to about 460 clock cycles), and a Multi-rate FIR [3] (270 tap low-pass FIR and 75 tap high-pass FIR, anti-aliasing FIR filters > 90 dB out of band, Decimation factor = 6, corresponding to about 460 clock cycles).
The measurements were performed separately for left and right channels of the car $H_l(f)$ and $H_r(f)$, and the $H_{inv}(f)$ was computed for each channel, averaging the absolute value of H_l and H_r, with only the dummy head in the cockpit. This results in a great robustness with respect to the number of people in the car and to their position.
Fig. 18 shows the measured SPL in the car with and without WFIR equalization, with a different number of people.
Fig. 19 shows the comparison among the three different kind of equalization filters. The target frequency response was a flat shape, except for the low-pass part of frequency at which equalization is not active. The results is not exactly flat, because of dynamic compression and median filter operation, (see section) but is still nice for listeners, see section 2.
Listening test were performed too, that confirm that there is a slight difference among the different filter structures, fig. 20.

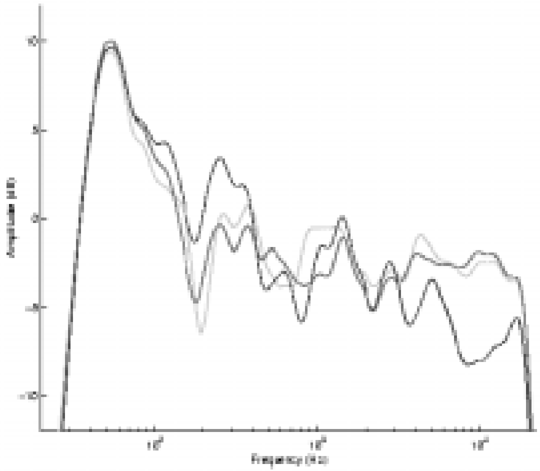


Figure 18 Car SPL with WFIR equalization (light gray) , with a passenger (gray), and without (black).

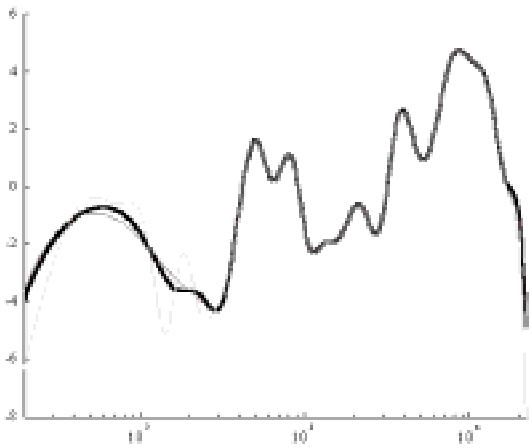


Figure 19 DSP WFIR equalization **(**black line**)**, Multi-rate FIR equalization **(**light gray line**)** and FIR equalization **(**gray line**)**.

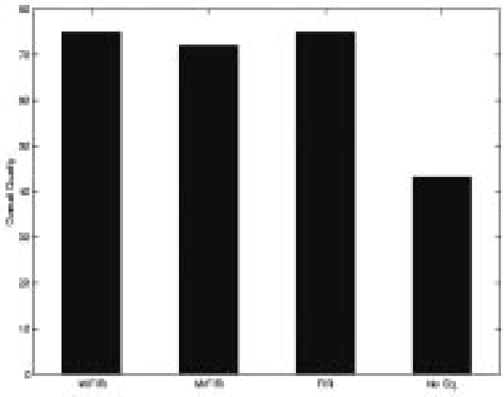


Figure 20 Listening tests, comparison between the three different filters (score 0-100).

Conclusion

In this paper an equalizing system for car cockpits was presented and validated with several listening tests and experiments. It was designed and implemented on a low cost DSP-based board. An automatic procedure for the synthesis of the filter coefficients was designed, starting on standard acoustic measurements.

Adopting the WFIR filter structure the flattening of the amplitude of the car frequency response is nice, and from the subjective point of view the sharpness and quality of the human voice reproduction is pretty increased and the frequency response in the low frequency range is very nice.

WFIR are useful tools in improving sound quality and operate pretty fine in any car position. However from the subjective point of view the difference among FIR, Multi-rate FIR and WFIR is slight, as long as they perform the same equalization.

The key point in car equalization is the choice and accuracy of measurements and its post processing. This issue affects the effectiveness of the equalization much more than the filter type. The direct comparison among WFIR, Multi-Rate and FIR filter shows that:

- the WFIR filter is the optimal choice for low frequency equalization, less than 150 Hz and very low filter length, less than 100 clock cycles;
- the FIR filter is optimal for high frequency equalization, medium filter length, 16 bit fixed point DSP, coupled with 2nd order double precision IIR for equalization of frequencies less than 150 Hz if needed;
- multi-rate FIR is the optimal choice for low frequency equalization, with high filter length (more than 300 clock cycles).

A prototype digital equalizing board was designed according to these considerations, and experimental results performed on commercial cars confirm that this technique is attractive for the next generation of car sound system.

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Alberto Bellini was born in Forlì, Italy, on December 18, 1969. He received the Dr. Eng. degree in electrical engineering in 1994 and the Ph.D in Electronics and Computer Science in 1998, both from the University of Bologna, Italy.
Since 1997 he is at the University of Parma, Italy, where currently he is an assistant professor of electrical engineering.
His currents research interests include Digital Signal Processing, non-linear modeling and control of complex systems, and algorithms for fuzzy and neural systems.

Gianfranco Cibelli was born in Reggio Emilia, Italy, on August 3, 1973.
He received the Dr. Eng. degree in Electrical Engineering at the University of Parma in April 1998. In 1999 he joined ASK Industries. His research interests include signal processing and acoustics.

Enrico Armelloni was born in Piacenza (Italy), in 1971. He received the Laurea degree in electrical engineering from the University of Parma in 1999, with a thesis about implementation, on DSP, of Warped FIR digital filters (WFIR), based on Frequency Warping technique.
From December 1999 to March 2000 he collaborated with ASK Industries to make experimental validation of warping filters inside car cockpits.
In July 2000 he collaborated with Ambiophonics Institute, to develop a Stereo-Dipole system, based on WFIR filters, implemented on DSP.
Since August 2000 he is a researcher at the Dipartimento di Ingegneria Industriale of the University of Parma. Main research activities are about technique for spatial equalization of sound and Active Noise Control for reduce noise from industrial machinery.

Emanuele Ugolotti received a Degree Laurea in Electronic Engineering from the University of Parma in 1992, with a final thesis work on physical modeling and practical implementation of electromagnetic brake for high power loudspeakers. Since then he was employed in R.C.F. S.P.A., where he designed electroacoustical transducers, especially compression drivers, horns and electronic sound processors as well. In January 1996 he moved to ASK automotive industries GROUP, as electroacoustics researcher. His current research interests are in the fields of on car interior sounding, in order to optimize the transducers-environment linking, and to optimize the sound field conformation.
He currently works in designing active subwoofers for first equipment car application, studying electronic correction of loudspeaker's performance and in sound car interior simulating by means of both auralization technique and boundary elements approach.

Angelo Farina received the engineering degree at University of Bologna in 1982. Ph.D in Technical Physics at University of Bologna in 1987.
Researcher at the University of Bologna since 1 November 1986 and at the University of Parma since 1 march 1992. Member of the Audio Engineering Society.
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