EXPERIMENTAL VALIDATION OF EQUALIZING FILTERS FOR CAR COCKPITS DESIGNED WITH WARPING TECHNIQUES

Alberto Bellini[1], Gianfranco Cibelli[2], Emanuele Ugolotti[2],

Filippo Bruschi[2], Angelo Farina[3]

[1]Dipartimento di Ingegneria dell'Informazione, University of Parma,
Parco area delle Scienze, 181/A, I-43100 Parma, Italy
Tel. +39 0521 905831, Fax. +39 0521 905822
email: alberto@ee.unipr.it
[2] ASK Industries S.p.A., via F.lli Cervi, 79,
I-42100, Reggio Emilia, Italy
Tel. +39 0522 388311, Fax. +39 0522 388322
email: cibelliG@askgroup.it, ugolottiE@askgroup.it, bruschiF@askgroup.it
[3]Dipartimento di Ingegneria Industriale, University of Parma, Italy
Tel. +39 0521 905854, Fax. +39 0521 905705
email: farina@pcfarina.eng.unipr.it

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. To accomplish this task the inversion of the measured Sound Pressure Level (SPL) should be performed. The inversion of this system transfer function is the key point of the equalizing procedure, and many advanced techniques were developed to this aim. However they produce a large number of FIR filter taps, and thus cannot be implemented real-time on a low cost DSP. "Frequency warping" theory allows to design a filter especially fitted for low frequencies with a small number of taps. In this paper an automatic tool to develop warped filters for target car cockpits equalization was designed and validated through experiments in commercial cars.

I. 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 it 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, which allows the real-time computation of a few hundreds of taps at 48 KHz. 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 still obtaining a fair equalization, which can be implemented on a low-cost DSP (3). This can be achieved for example exploiting fast convolution techniques for discrete-time signals, like the 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 the use of 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.

II. FREQUENCY WARPING THEORY

The basic idea of the warping theory is well represented in the following chart. In fig. 1 the FIR-like structure (WFIR) is reported (4), while on fig. 2 is reported a simplified structure adopted for efficient implementation.

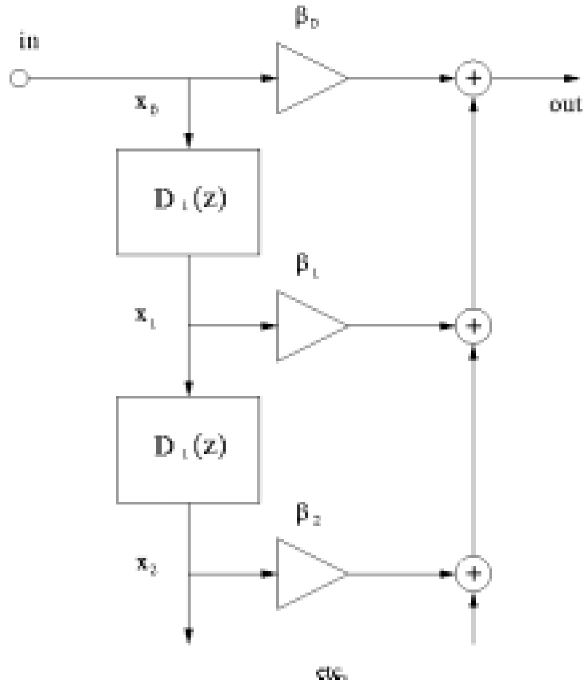

Figure 1 WFIR filter structure general structure.

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

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

that allows to transform the z plane into the "warped" $D_1(z)$ plane. λ is referred to as the "warping parameter" and it ranges from -1 to +1, $D_1(z)$ is the warped delay unit. Since it is frequency dependent, the frequencies are remapped so that the resolution increases for the low part of the spectrum, with a positive value of λ, and the other way round for negative values of λ, still keeping the sampling rate constant. This transformation is consistent with the logarithmic shape of human hearing system resolution. Fig. 3 shows the effect produced by the bilinear transformation $D_1(z)$.

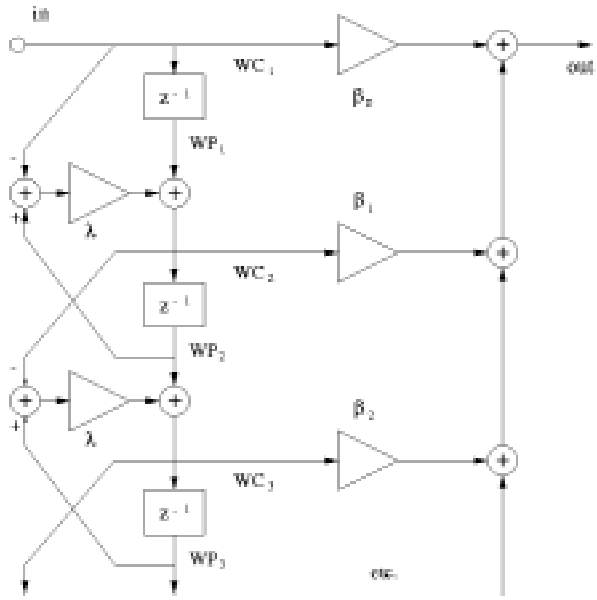

Figure 2 WFIR filter structure basic blocks.

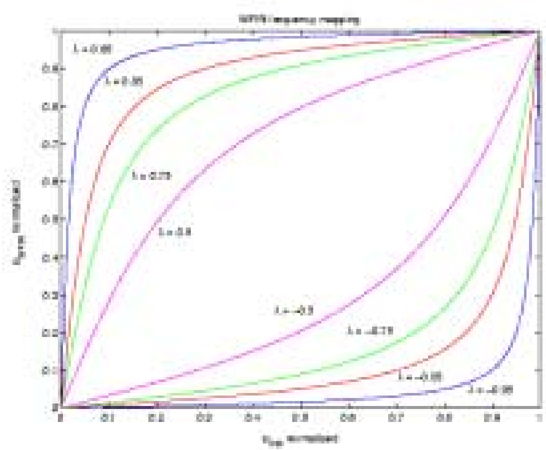

Figure 3 Bilinear transformation of frequency with different λ values

The choice of the value of λ allows tuning the “stretching of frequency”, 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 48 kHz, λ is around 0.8.

Advantages and drawbacks of warped filters are addressed by their specific differences versus “classical” filters. In a classical filter the frequency resolution is constant in all the frequency range, fig. 4. However since human hearing sensitivity is so that frequency resolution is of about one third of octave, the equalization accuracy should be sharp at lower frequencies and weak at higher. This 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.

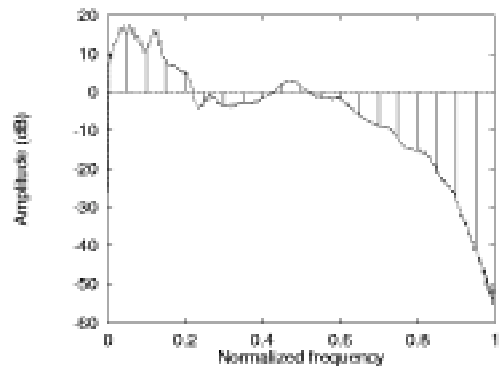

Figure 4 SPL shape of a target environment.

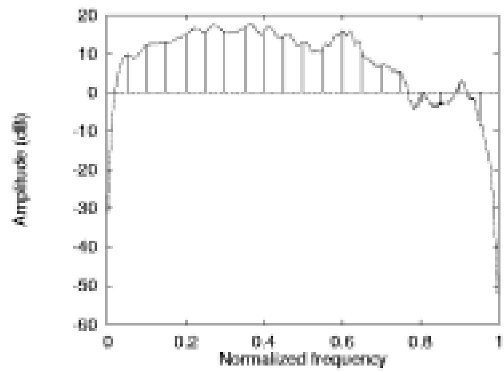

Figure 5 Pre-warped SPL ($\lambda < 0$).

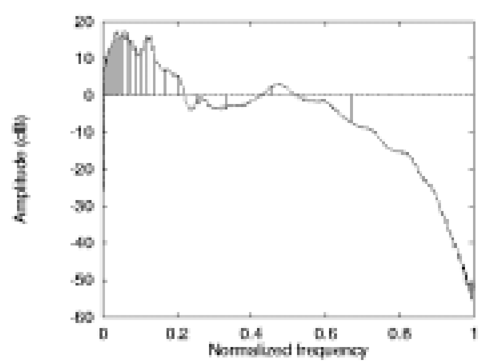

Figure 6 SPL synthesized with a WFIR.

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 a new spectrum is obtained, see fig. 5. At this step the shape can be operated by the same WFIR structure as before but with an opposite value of 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 produces 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 from five to ten times more than 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 the impulse response coefficients can be kept without affecting the accuracy of the low frequency reproduction.

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

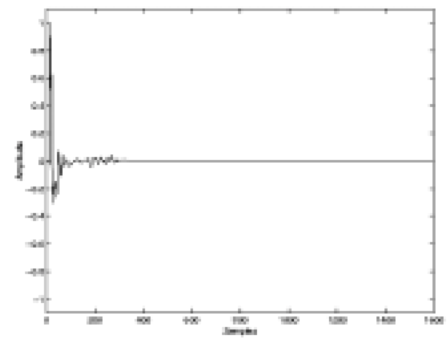

Figure 7 Loudspeaker impulse response measured in an anechoic chamber.

It also represents the taps of the FIR filter that can approximate the loudspeaker behavior. If we 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.

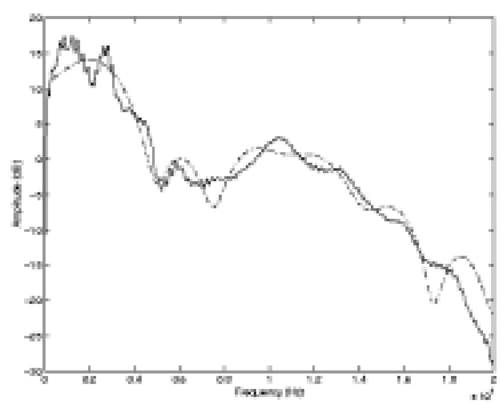

Figure 8 Amplitude of the loudspeaker frequency response before (solid line) and after the truncation (dashed line).

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

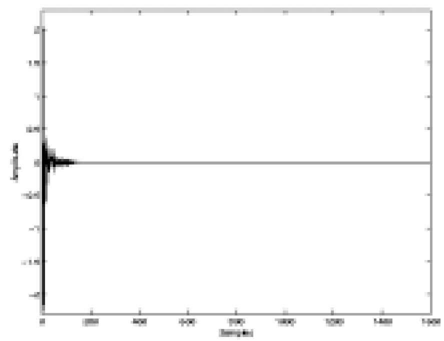

Figure 9 Warped Loudspeaker impulse response, measured in an anechoic chamber

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. On the other side, the high frequency part is completely different from the original one. This example clearly shows the effect of the application of warping transformation to frequencies.

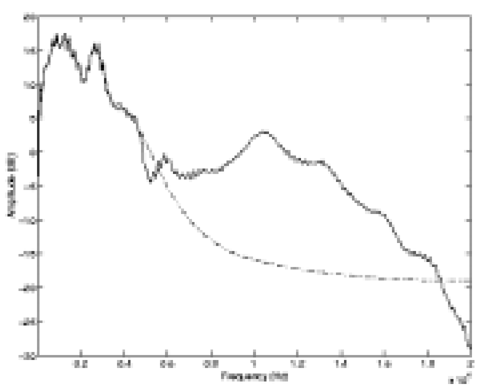

Figure 10 Amplitude of the warped loudspeaker frequency response before (solid line) and after the truncation (dashed line).

III. CAR EQUALIZATION

The aim of equalization is to obtain an overall flat band frequency response. A first option is that of adding a digital system to the audio source, characterized by a system transfer function $H_{inv}(f)$, so that the frequency response of the overall chain is linear. This requires the assumption of a linear system, which is not the case due to loudspeaker intrinsic structure. Specifically the basic car audio system is usually a two or four-channel audio system, and in the following it will be assumed a two-channel linear system. This means that the equalization should be applied only to the part of the signal spectrum where non-linear distortion is not significant. This can be made applying equalization filters only to the mid-range of signal spectrum, or keeping out from the digital equalization chain those loudspeaker that are mainly affected by distortion, like woofers or sub-woofers.

The adoption of a filter which realizes directly $H_{inv}(f)$ is not the best solution, and can lead to unexpected drawbacks, as an increase of the overall distortion, since a non-accurate correction of frequency spectrum can modify the spectrum in the opposite direction of what is desirable. Therefore psychoacoustics considerations on human hearing 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: fig. 11 is an example of the optimal target shape for a commercial car cockpit. In general the target shape is dependent on the specific target car considered.

From human hearing resolution it stems that is ineffective to equalize all the frequency range with the same resolution. In fact at low frequencies the human hearing system can distinguish different sounds which are effectively near in frequency, while at high frequencies even the most sensitive ear fails. The equalization of very narrow peaks or holes in the frequency response would be inefficient and even unpleasant.

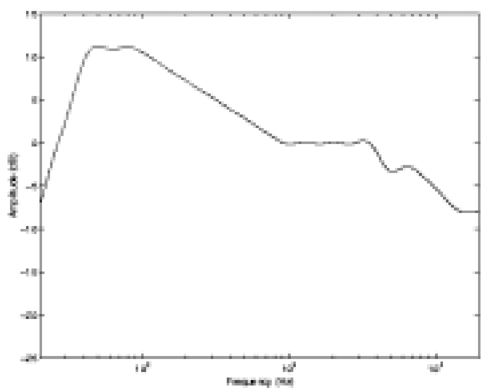

Figure 11 Target frequency response shape.

Therefore to reproduce human hearing resolution the inverse harmonic response $H_{inv}(f)$ was processed by a median filter (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. 12.

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 for instance to ±6 dB, so that the correction would be pleasant, smooth and natural.

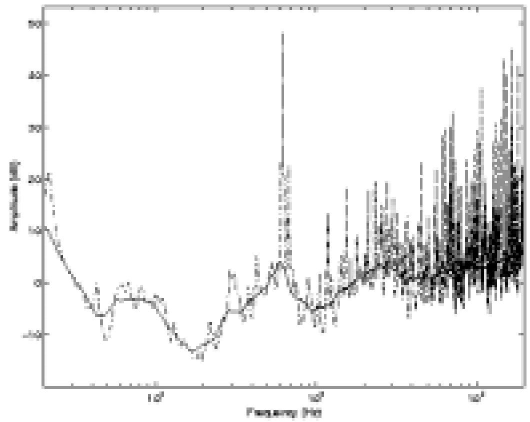

Figure 12 Amplitude of $H_{inv}(f)$ (dashed), and its processed version (solid).

IV. 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, fig. 13.

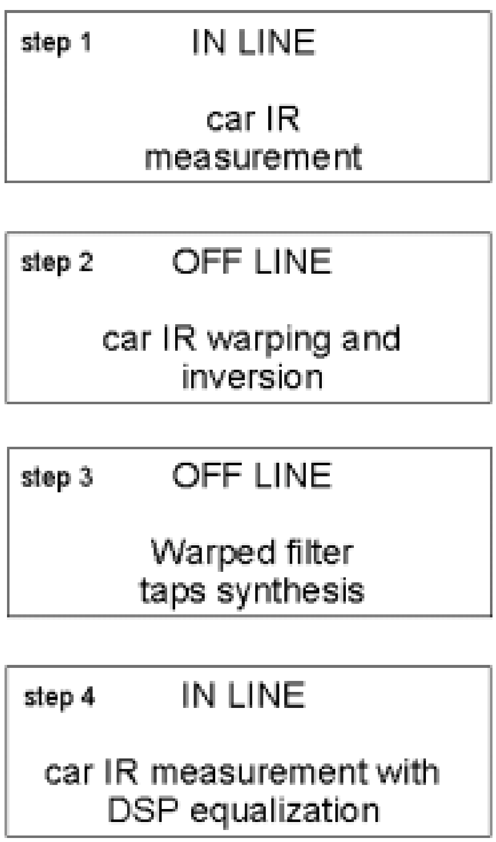

Figure 13 WFIR automatic design flow-chart.

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). 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 directly obtained applying the analytical transformation reported in the previous section to the signal.
3. Synthesis of the taps of the "warped" filter.
4. Measurements of the impulse response on the target loudspeaker with the DSP equalization.

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). And this structure was implemented exploiting all the potentiality offered by the DSP platform adopted. In these experiments the AD SHARC 21065Ldevelopment board was used, together with the AD1819A stereo CODEC at 48 kHz.

Every step can be realized in different ways in order to implement different approaches. In facts many kinds of impulse response (IR) are feasible as well as many IR inversion, as many different architectures for WFIR implementation.

Therefore a general multi-step procedure was designed, consistent with the flow chart shown, where many degree of freedom are left at each step. So, depending on the specific car cockpit considered, each box is characterized in order to obtain the most efficient equalization, see fig. 14.

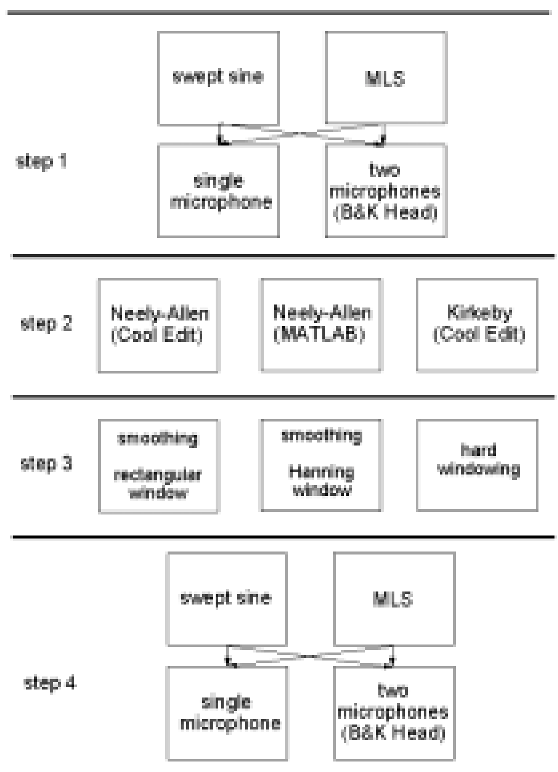

Figure 14 WFIR multi-choice automatic design flow-chart.

The term “in line” means that the operations of the related step are to be performed in real-time, i.e. in compliance with the adopted sampling-rate. On the other side “off-line” operation can be performed without time constraints, and on an host computer where previously acquired data are processed.

In fig. 14 the four steps flow chart is exploded in order to show details about different approaches. As for the step 1 the measurement can be made by means of swept sine or (Maximum length signal) MLS techniques; each of these methods can be implemented using one single microphone or two binaural microphones included in the B&K Dummy Head and Torso Simulator, see fig. 15. The former choice varies the input signal fed by the audio source to the microphones.

Figure 15 Target car cockpit. Dummy head in the driving position for acoustic measurements.

As for step 2 at first the warping of the measured IR is perfomed, and then the obtained frequency response is inverted. This latter operation can be done using either Neely-Allen technique (modulus inversion, (8)) or Kirkeby technique (modulus and phase inversion using regulation parameter, (9), (10)). Each of them are realized by means of the Aurora plug-in software (11) inside Cool Edit Pro and/or Matlab.

As for step 3 the selection of the filter taps to be used for implementation is made relying on a smoothing operation together with a windowing one. A rough rectangular windowing (box car) without smoothing generates a filter that equalize only low frequencies in agreement with the theory of frequency warping. As for step 4 the same options as in step 1 are available.

Several commercial cars were used as the target environment. The acoustic measurements performed within the car cockpit by means of Aurora (11) 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.

All the experiments were made comparing results with and without digital equalization according to the structure depicted in fig. 16.

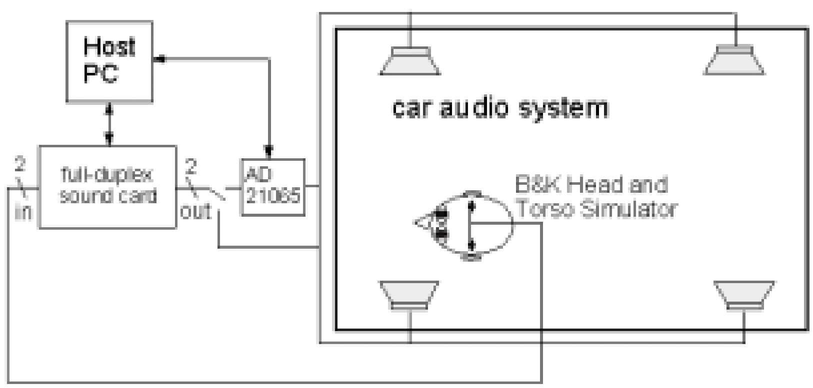

Figure 16 Experimental tests structure.

The target car was an Alfa 156, equipped with a specifically designed sound system, made of 9 loudspeakers and of a 7-channels amplifier. The digital equalizer was a WFIR with 89 coefficients, corresponding to about 460 clock cycles, λ =0.87, and to a FIR with 451 taps.

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 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. The target shape was almost constant at frequencies higher than 100 Hz, and no equalization was applied below.

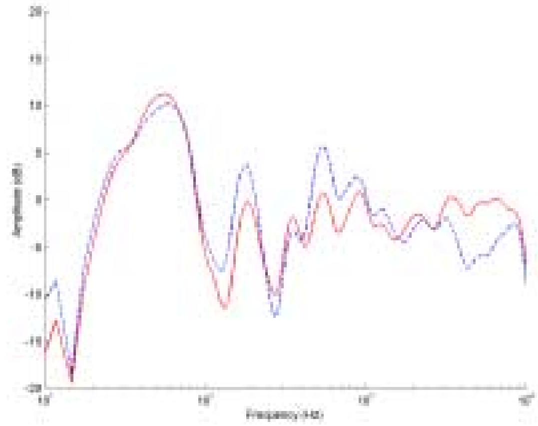

Figure 17 Car SPL with WFIR equalization (solid line) and without (dashed line), measurements performed with the B&K Dummy Head and Torso Simulator and two microphones (sub-woofer disabled).

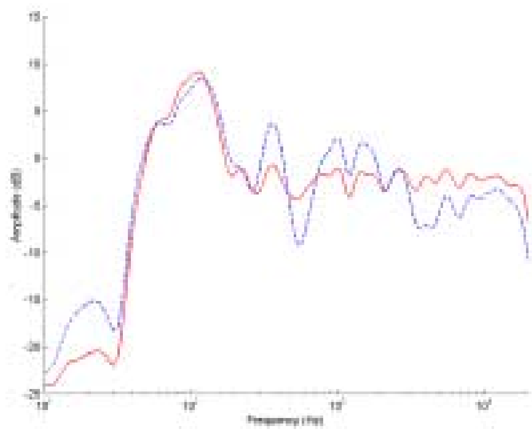

Figure 18 Car SPL with WFIR equalization (solid line) and without (dashed line), measurements performed with a single microphone.

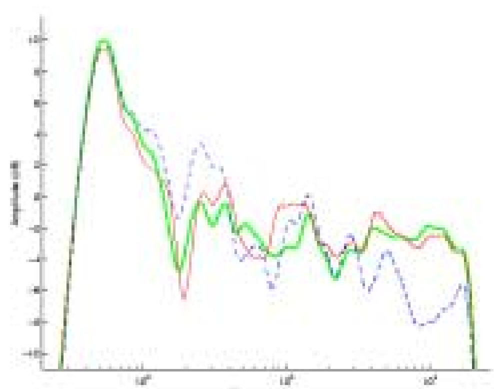

Figure 19 Car SPL with WFIR equalization with a different number of people (solid line) and without (dashed line), measurements performed with the B&K Dummy Head and Torso Simulator and two microphones (sub-woofer enabled).

Fig. 17 and fig. 18 show the comparison of SPL measured with and without DSP equalization when the sub-woofer is disabled, in order to avoid the influence of high distortion transducers.

Fig. 19 reports the comparison of SPL measurements with and without WFIR equalization when the sub-woofer is active. In this chart the SPL is measured twice with the insertion of WFIR equalization, with one and two passengers. The slight difference that occurs shows how the proposed equalization system is pretty robust with the changes in the acoustic configuration of the target environment.

From an objective point of view the results obtained with a single microphone (fig. 18) look better than the ones obtained measuring the SPL with a B&K dummy head with two microphones. Listening tests however show that the latter is more effective for the human hearing system. The SPL shape is pretty flat in the former case just because the system is more linear and it is easier to reproduce ideal condition. On the other side with two microphones a heavy smoothing must be applied in order to compensate properly averaging of channels, and it looks bad from the flattening point of view, but the listeners appreciated it.

Listening tests show that the results obtained with WFIR equalization is sensibly better than the normal car audio system, and it is similar to the results obtained with a standard FIR with the same computational costs. However WFIR equalization is much more effective at low frequencies, and seems a nice trade-off in those cases where a limited number of coefficients is available, and where low frequency reproduction is a key issue.

V. CONCLUSION

In this paper an equalizing system for car cockpits based on Warped filters 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. The optimal choice of them is dependent on the target environment and on the desired characteristics. 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, for implementation on 16 bit fixed point DSP.

The use of dummy heads with two microphones seems a nice choice as long as the two channel measurements are averaged and properly smoothed, in order to obtain a nice and robust equalization apart from listeners' position.

Acknowledgments

This paper was supported by EC under the ESPRIT project APLODSP 28.5873.

REFERENCES

(1) W. Klippel, "Compensation for Nonlinear Distortion of Horn Loudspeakers by Digital Signal Processing", Journal of the Audio Engineering Society, November 1996, vol. 44, n. 11, pp 964-972.

(2) A. Bellini, G. Cibelli, E. Ugolotti, A. Farina and C. Morandi, "Non-linear Digital Audio Processor for dedicated loudspeaker systems", IEEE Trans. on Consumer Electronics, August 1998, pp. 1024-1031.

(3) G. Cibelli, A. Bellini, E. Ugolotti, A. Farina, C. Morandi, "Experimental validation of loudspeaker equalization inside car cockpits", preprint #4898 of AES 106th convention, Munich, Germany, May 1999.

(4) M. Karjalainen and E. Piirilä and A. Järvinen, "Loudspeaker Response Equalisation Using Warped Digital Filters", Proc. of NorSig-96, September 1996, Espoo, Finland, pp. 367-370.

(5) E. Zwicker and H. Fastl, “Psychoacoustics-Facts and Models”, Springer, Berlin, 1990.

(6) J. O. Smith and J. Abel, “The Bark Bilinear Transform”, in Proc. of IEEE ASSP Workshop, 1995, Mohonk, New Platz, NY.

(7) S.K. Mitra and J.F. Kaiser, “Handbook of digital signal processing”, McGraw-Hill, New York, 1993.

(8) S. Neely and J. Allen, “Invertibility of a room impulse response”, Journal of Acoustic Society of America, no. 66, pp. 165-169, 1979.

(9) O. Kirkeby, P. A. Nelson, H. Hamanda and F. Orunda-Bustamante, “Fast deconvolution of multi-channel systems using regularization”, ISVR Tech. Rep. of Univ. Southampton UK, no. 255, 1996.

(10) O. Kirkeby, P. Rubak and A. Farina, “Analysis of ill-conditioning of multichannel deconvolution problems”, Proc. IEEE Workshop on Application of Signals Processing to Audio and Acoustic, pp. 155-158, Oct. 17-20 1999.

(11) A. Farina and F. Righini, “Software implementation of an MLS analyzer, with tools for convolution, auralization and inverse filtering”, Preprint of the AES 103th Convention, New York, September 1997.