Experimental validation of warping filters inside car cockpits

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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 psychoacoustic 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]. 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 which 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.

The direct inversion of the measured car frequency response however is a difficult task, since it usually requires a high number of taps, and therefore it is not suitable for a real-time implementation. Therefore several simplifications must be done to obtain a fair equalization, which can be implemented on a low-cost DSP [2]. Another options is that of exploiting signal processing techniques to realize a better filter implementation. To this aim warping theory can be exploited. It is basically a bilinear transformation in the z-transformed domain, which stretches the frequency range according to the diagram in fig. 1-left, so that a larger number of coefficient is used for the low frequency, and less for the higher frequency. As an example in figs. 2 is clearly shown the comparison with the different resolution that can be obtained after the "warping" in the case of a car cockpit acoustic response. Moreover this transforma-

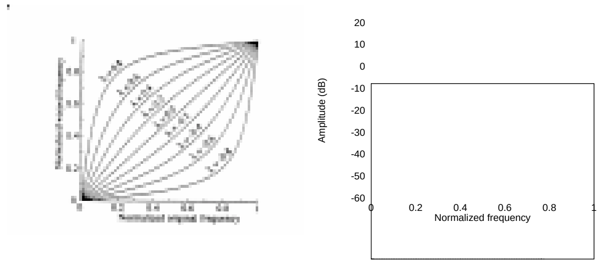

Fig. 1. Bilinear transformation of frequency with different λ values (left), SPL shape of a target environment (right).

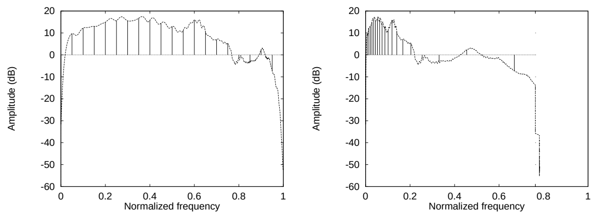

Fig. 2. Pre-warped SPL ($\lambda < 0$) (left), SPL synthesized with a WFIR (right).

tion is consistent with the logarithmic resolution of the human hearing system. Experimental results show that a very low number of FIR filter taps is sufficient to reach an effective equalization.

Experimental results

The warping theory was exploited to realize a modified FIR structure. This structure, reported in fig. 3 is usually referred to as WFIR. 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, [3].
3) Transformation of the impulse response in the "warped" domain. This can be made using the WFIR

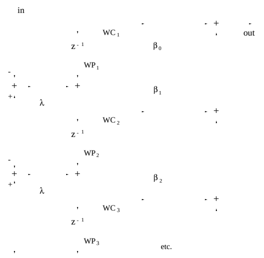

Fig. 3. WFIR filter structure.

structure reported in fig. 3. This requires the β_i coefficients of the original impulse response, and the application of -λ. 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 (see fig. 2-left).
4) Synthesis of the taps of the "warped" filter. To this aim the LPC function provided by Matlab was used, which is an all-pole (AR) structure. Specifically the impulse response of the warped filter obtained at the end of the previous step was approximated by an all-pole polynomial function, whose order is the same of the "warped" filter. The polynomial function coefficient are then the taps of the filter which approximate the SPL of the target car cockpit.
6) Synthesis of the car cockpit inverse filter (see fig. 2-right). This can be simply accomplished swapping numerator and denominator of the polynomial function. Therefore the final structure of the warped filter, which implements SPL^{-1} is the same as fig. 3, where β_i are the taps obtained with the above mentioned procedure and λ is the optimal warping coefficient (with positive sign).

A Citroën Evasion was used as a target environment. The acoustic measurements performed within the car cockpit by means of Aurora [4] 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 increase with the order of the filter. It can also be noted that a very low number of taps is sufficient to obtain nice results. A DSP development board provided by Analog Devices was used as the hardware platform. It relies on the AD21061 DSP. All experiments were performed at 44100 Hz sampling rate, as stated by the current Audio CD standard.

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 350 taps in terms of computation complexity. In fig. 4 the frequency response within the car for both the channels are shown with and without the digital FIR equalization. In figs. 5, 6 the frequency response within the car for both the channels are shown with and without the digital WFIR equalization. Experiments show that the equalization accuracy especially at low frequency is much better with the WFIR filter, while the computation complexity is similar.

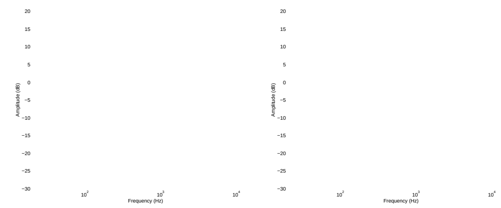

Fig. 4. Frequency response with 350 FIR equalization (solid line) and without (dashed line), Citröen Evasion cockpit, left channel (left), right channel (right)

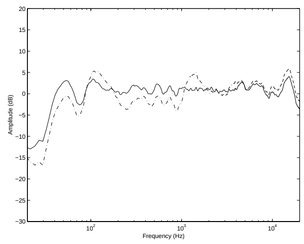

Fig. 5. Frequency response with 27 WFIR equalization (solid line) and without (dashed line), left channel.

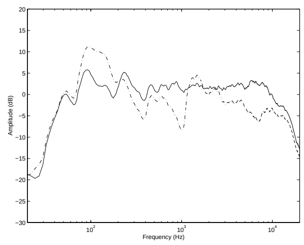

Fig. 6. Frequency response with 27 WFIR equalization (solid line) and without (dashed line), right channel.

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