APLODSP, DESIGN OF CUSTOMIZABLE AUDIO PROCESSORS FOR LOUDSPEAKER SYSTEM COMPENSATION BY DSP

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Abstract – This paper summarizes the results of the ESPRIT project APLODSP. The goal of this application experiment is to develop adequate models for the simulation of the non-linear behavior of loudspeakers and to design a dedicated audio processor to reduce sound distortion. This involves the definition of a systematic design flow for anti-distortion audio processors, and the effective exploitment of CAD tools for the automatic implementation of the defined algorithms. The audio processor will be implemented with a DSP, using state-of-the-art tools for simulation, validation and synthesis. DSP is the emerging low cost technology for audio processing, and in particular for car-audio systems. In fact, car manufacturers are planning to reduce the cables inside the car and to use a single cable to distribute the main signals multiplexed all around the vehicle. This transition to digital audio signal transmission will foster the use of active loudspeakers, equipped with dedicated digital audio processors. The audio processor, designed and tested within this ESPRIT project, can be seen as a first step in this direction. It shows how the loudspeaker distortion can be reduced by digital signal processing, and it exploits the versatility of digital designs in order to allow hardware re-use for different car models. It also allows a very fast redesign to fit many different purposes.
A semi-automatic design flow for the design of anti-distortion audio processors is available, which synthesizes a dedicated audio processor for each loudspeaker model, once suitable loudspeaker parameters have been specified.

I. INTRODUCTION

The compensation of loudspeaker distortion is a difficult task, since many non-linear phenomena occur and affect the behavior of the transducer. Two possible approaches can be adopted: an open-loop or a closed loop correction structure. The latter requires sensors able to detect in real time the electric and mechanical quantities directly from the transducer. The sensor may be either a microphone or a current sensing and in both case non-repeatability and cost are the major drawbacks. The former requires an accurate parametric model of the loudspeaker which allows to design a dedicated audio processor, once the main loudspeaker acoustic parameters are provided. The audio processor such obtained can be programmed and inserted before the transducer in order to enhance sound reproduction quality. This was the choice adopted in the APLODSP project, since we can rely on the loudspeaker non-linear model depicted in (1), and simulated with Synopsys COSSAP (2), which is a fine trade-off between generality, effectiveness and cost. This allows the semi-automatic design of a dedicated audio processor for each specific acoustic target.

The audio processor which was designed in the ESD Application Experiments is basically composed by a two dimensional filter structure where the coefficients depends on the actual cone displacement. The implementation phase requires therefore a synthesis tool which can compute the suitable coefficients for a dedicated loudspeaker and an efficient architecture for FIR convolutors. In this framework two solutions to increase the performance of the FIR implementation were investigated. The former, based on Warping theory (3), performs the stretching of the signal in the frequency domain, and aims at syntesizing more accurately the low frequency part of the filter, while the latter is a fast convolution technique which allows to lower computational cost, when filters with a very high number of coefficients are required. Moreover given the features desired as a function of frequency Multi-rate filters were analyzed too. They can be a nice trade-off between performances and costs, since the frequency spectrum is often split in parts with different requirements.

All the different algorithms were included in a dedicated software tool, which automatically computes the filter coefficients and code, starting with standard acoustic measurements performed in the target environment.

Moreover a solution to increase sound harmonization is proposed (4). It increases the global quality of reproduction in such a hard environment as a car cockpit, in order to mask distortion phenomena.

Several experimental results were performed in listening room for the non-linear filter and in car cockpits for the equalizing filters, in standard and warped forms. In this paper they are presented and discussed, in order to show the effectiveness of the proposed audio processor.

Commercial development tool by Analog Devices and Texas Instrument were used as implementation platform, and the COSSAP xdcg tool was used to obtain automatic generated DSP code.

II. PROJECT OBJECTIVES AND WORPLAN

The project was developed in four main tasks: the task 1 was devoted to the definition of an effective loudspeaker model, the task 2 with the design of an effective architecture for distortion compensation, realizable in digital technology with a parametric audio processor, the task 3 with the implementation and validation of the audio processor, the task 4 with reporting and dissemination of results.

Task 1 was mainly concerned with the definition of a non-linear model for loudspeakers, model that is needed to perform adequate simulations of its non-linear behavior. A special reference was made to low-frequency loudspeakers, usually referred to as woofer, since they are more affected by non-linear phenomena.

Technical literature was studied in order to seek for the state of the art in the field of non-linear model, with particular attention to loudspeaker systems analysis. The common Small-Thiele approach with its circuital equivalent was adopted, but it was extended in order to achieve an accurate modeling of non-linear behavior.

A non-linear model for low frequency loudspeaker system was defined and validated versus experimental measurements of commercial loudspeaker systems, by means of the COSSAP analysis tool.

The use of current driving circuit for loudspeaker was investigated too, in order to verify the potentiality of this relatively unexplored approach.

In task 2 the architecture of the audio processor was designed and simulated with COSSAP tools. It is an open loop structure that relies on the non-linear model defined in the previous task. It is basically composed by a two-dimensional filter structure, where the coefficients change at each time step, depending on the actual cone displacement. The acquisition of EDA tools allowed a strong increase in design potentiality.

In this task the following items were thoroughly analyzed. Design and simulation of a parametric non-linear inverse filter, based upon the model described in the task 1, was performed by means of COSSAP tools. In order to ease the synthesis of an efficient architecture, some solutions were tackled, in order to increase computational efficiency, and account for human hearing resolution, which is related to octaves.

Moreover some solutions for the increase of sound harmonization were analyzed. They aim at reducing the cross-talk effects, i.e. the influence of a stereo channel on the other, and other spurious acoustic effects. To this purpose stereo-dipole systems were analyzed.

Another issue is that of the target frequency response shape after the pre-elaboration performed by the audio processor. It was noticed that a sharp flat-band response is not desirable, as the anechoic chamber sound reproduction reduces natural sound contents and richness. Therefore it was investigated the adoption of a specific target frequency response curve, whose shape increase low frequency contents, as required to reduce running car noise. Of course this solution is especially fitted to car cockpits.

The COSSAP tools are an effective way of simulating digital architecture that will be implemented on a DSP platform. The library provided is composed by a large number of predefined blocks for each specific field of interest, and many dedicated blocks (like FFT, FIR, IIR) are useful to the purposes of the project. It provides lots of automatic procedures to implement the architecture on a DSP, without handwriting code, or a way to design dedicated hardware structures, producing directly the VHDL code. These advantages can be obtained at the price of an expensive license and training of designers.

In task 3 the audio processor for car cockpit equalization and harmonization was implemented on a commercial DSP evaluation board. To this aim the AD SHARC 2106x boards were adopted. The structure designed in task 2 was directly mapped on the DSP platform with the following specific issues.

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.

To this aim the extensive use of simulations allows the definition of the best trade-off between architecture complexity and overall efficiency. Warping theory and fast convolution techniques were studied to gain from their potentiality.

On the hardware point of view on the other side many different commercial DSP-based platform were used as a prototype. Specifically SHARC Ez-Lite and Bittware Spinner, AKM7712. FPGA-based implementation feasibility was also considered, but due to the unavailability of user-friendly hardware development kit, already set-up with proper interfaces (audio CODECS, etc), this possibility, though theoretically promising, was not practically adopted at this time.

Several measurements procedure were also analyzed and compared. Specifically is of key importance the choice of the input signal to be used to obtain the acoustic characterization of the target car cockpit. To this aim Cool-Edit pro and the Aurora plug-in were used to adopt MLS signal as stimuli for the proper frequency response characterization. Another option is the use of a swept-frequency sinusoidal signal to excite all the different frequency at different time.

Another key point is the separation of channels, left and right or front and rear of car can be used separately, and they can be inverted for equalization.

Moreover in the experiments it was noticed that the choice of the inversion strategy could affect in a strong way the overall equalization efficiency. To this aim several methods proposed in the literature were compared.

The choice of the hardware platform is a critical issue, since it can affect the whole automatic design flow. Some platform in fact are more fitted to the COSSAP tools, however the adopted solution was that of exploiting the most common hardware tools in the audio market, to gain from experience of researchers and competitors, i.e. Analog Devices SHARC.

The direct inversion of the loudspeaker model has some drawbacks. First of all the dynamic range of the correcting filter should not be excessive in order to avoid deep holes in the frequency range which could lower the audio quality. Moreover a flat-band target frequency response is not desirable, since the audio quality tends to be decreased. Nobody would have an anechoic chamber as a listening room. And especially as far as car cockpits are concerned a dedicated comfort target frequency response shape was studied and validated by listening tests.

Tuning the agreement between the COSSAP simulation and synthesis tools and the hardware platform, in order to achieve an efficient and fast design implementation.

Task 4 was concerned in the dissemination of the progress achieved in the design flow of anti-distortion audio processors and to the definition of methods for measuring improvements in design capabilities of ASK.

Several papers were issued by the research activity of the ESPRIT project, and most of them were presented in high-level international conferences in the field: (1), (4), (5), (6), (7), (8), (9), (10).

A few demonstration of the DSP-based prototype were performed setting up some commercial cars for demos, as an example Fiat Lancia Delta, Citroen Evasion, Alfa 156, Honda Civic, Audi A6.

III. THE APPLICATION EXPERIMENT: AN AUTOMATIC DEVELOPMENT TOOL FOR THE DESIGN OF EQUALIZING FILTERS FOR CAR COCKPITS

The validation of the audio processor was made relying on three steps: automatic design of the audio processor coefficient, implementation, and in-car measurements.

Fig. 1 summarizes the whole procedure. It is composed by four main parts: acoustic measurements of the target car cockpits, performed by means of the Aurora plug-ins of Cool-Edit Pro (11); automatic synthesis of filter coefficients; DSP code synthesis, which is common to most options; acoustic validation of the target car cockpit with the insertion of the audio processor, which is basically the same as the first step. Each part features some options. As for the first it is possible to change the signal used for system characterization, as for the second the usual options coming from signal processing theory are available (12), (13), (14), as for the third the DSP code is quite general and allows switching from FIR filter to WFIR filter or Multi-Rate filters.

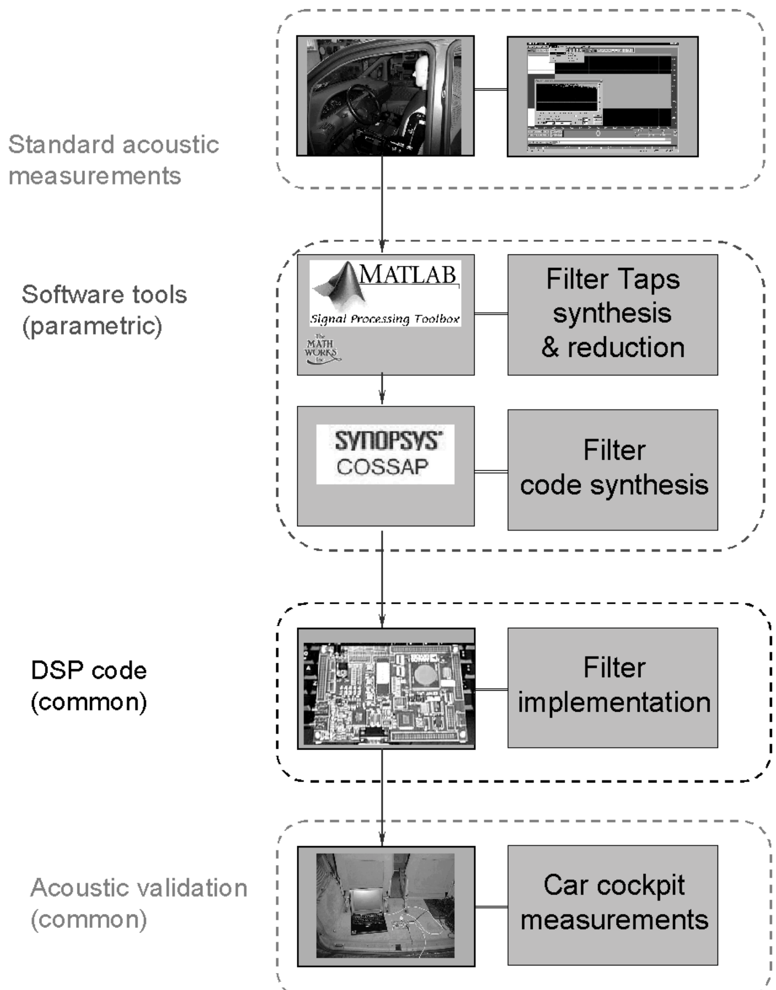

Figure 1 Automatic design flow for the audio processor coefficient.

This chart reproduces the actual use of the tools produced by this application experiments. At first the target car is acoustically characterized, then the measurements are processed, in order to obtain suitable filter coefficients, then the coefficients are mapped into the audio processor memory and the same car, with the insertion of the audio processor, is measured and compared from previous performance from both subjective and objective point of views.

The second step deals with the implementation, and a few options are made available to the user. In fig. 2 for instance the Multi-rate filter equalizer and the stereo-dipole system architectures are shown. Both are integrated in the audio processor.

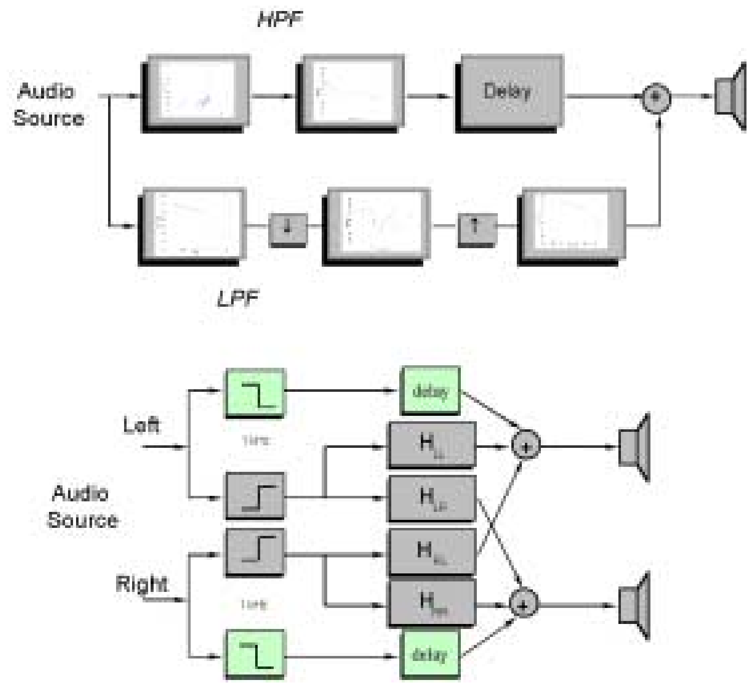

Figure 2 Architecture of the audio processor. Multi-rate filter based equalizer (top), Stereo-dipole system (bottom).

The last step is the experimental validation of the audio processor, which was made on several commercial cars, as listed in the previous section. Fig. 3 reports the SPL measurements made on a Fiat Lancia Delta. It can noticed how the audio processor allows to tailor the frequency response, just fitting the desired response, which was chosen by experts, in order to obtain an increase of sound comfort. In this experiment the multi-rate filter structure was adopted.

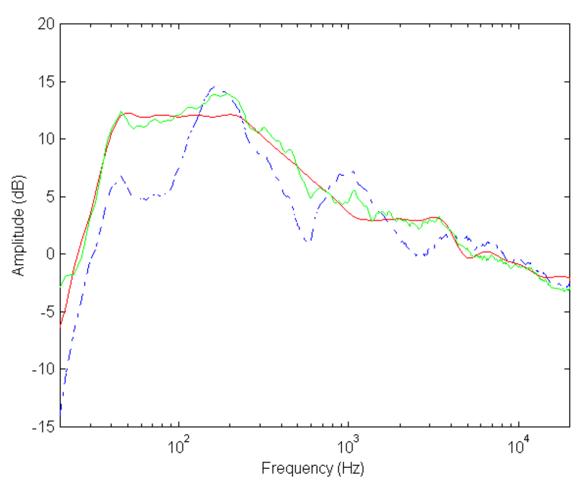

Figure 3 Car SPL with (solid line) and without (dashed line) Multi-Rate filter equalization, and target shape.

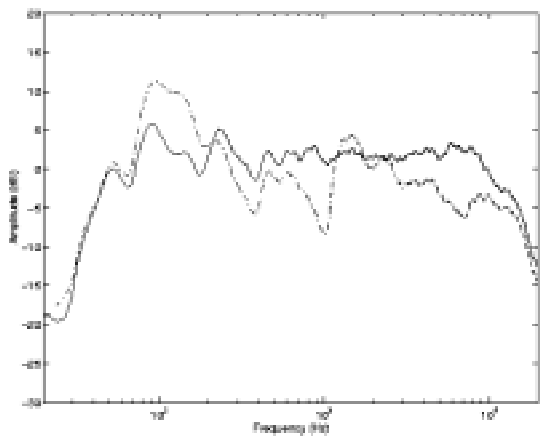

Figure 4 Car SPL with (solid line) and without (dashed line) 27 WFIR equalization.

Fig. 4 reports the SPL measurements made on a Citroen Evasion. It can be noticed how the audio processor flattens the frequency spectrum. Listening tests too confirm an increase in sound reproduction quality. In this experiments the WFIR filter structure was adopted.

Subjective tests were performed too, recording the answers of ten listeners to 7 questions, concerning sound quality, localization, and harmonization. The value reported is the mean score, where 8 correspond to the maximum evaluation (fig. 5). A general increase of the score occurs with pre-elaborated sound (empty histogram), and this confirms the effectiveness of the proposed approach. The test was composed by the evaluation of the following quality benchmarks: Q1 Initial sound sensation; Q2 musical scene location (voice and instrument position sharpness); Q3 Amplitude of sound front; Q4 Natural sound reproduction; Q5 Low frequency response; Q6 Medium frequency response; Q7 High frequency response.

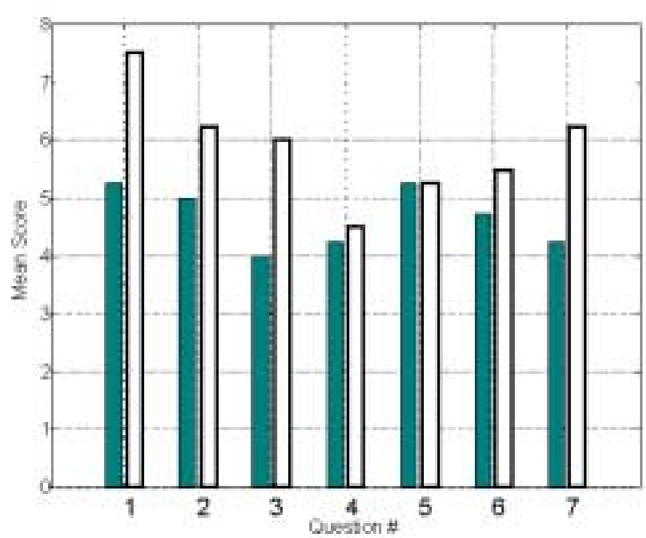

Figure 5 Subjective test results: stereo dipole score in empty histograms.

IV. LESSON LEARNT

In order to achieve an accurate modeling of non-linear behavior of loudspeaker systems commonly used Small-Thiele approach must be extended. The adopted solution was that of defining a two-dimensional filter bank. At first the easiest solution was to use the amplitude of the electric input signal as the selector for the choice of the bank to be used. But simulation and experiments show that, though more complicated, the use of the actual cone displacement, as a selector is more effective. In fact the non-linear behavior is directly related to the distance of the cone from its rest position, while the use of signal amplitude to obtain this information may be perilous.

The COSSAP tools require an event-driven project, in the sense that the links between the blocks are driven by the last block in the chain. At every simulation step the user can set how many data samples can be obtained from the previous block, thus adapting the design to real-time environment simulations. The simulation step states the sampling period and every other time information is referred to it. When specific target architecture is defined, actual information on the sampling rate are required. This last operation is the most interesting point, since it allows a very easy way to design real-time system with high-level blocks on DSP. The synthesis of the DSP code however is not fully automatic, since some adaptations are needed, depending on the target board used, on the amount of memory available and on the CODEC used. The addition of this information requires a great knowledge of the target hardware platform by the designer. This skill on the other side is required to obtain an effective hardware implementation.

MATLAB and its Signal Processing toolbox provide an excellent means to elaborate measurements data, for instance to smooth undesired peaks or holes in the frequency spectrum, which cannot be reproduced by a digital processor. Moreover they provide a powerful set of synthesis tools, which allow the computation of the filter coefficient, given the desired frequency response shape.

The choice of a suitable hardware platform, and especially of an effective combination of hardware and software tools, allows a dramatic reduction of design time and the achievement of nice performances. In this direction the adoption of SHARC based boards, coupled with common software tools, like Matlab and Cool-Edit has increased the performances, and reduced the design time. More complex design tools, like COSSAP, are much more effective in the simulation phase, and when planning the design of dedicated hardware structures.

The choice of the measurements to be performed for each car cockpit is a key point. An effective evaluation of the frequency response is a mandatory input for the equalization system. Careful analysis of the best choices of input transducers and stimuli, together with an effective inversion strategy, parametrical for each single target often allows great improvements of sound reproduction quality.

V. CONCLUSION AND FUTURE DEVELOPMENTS

The main goals of the application experiment were achieved. Suitable models for the simulation of the non-linear behaviour of loudspeakers were designed together with a dedicated audio processor in order to reduce sound distortion. The latter was coupled with a semi-automatic development tool that allows computing the optimal filter coefficients, given proper acoustic measurements of the target car cockpit. To this aim extensive simulation for the characterization of both algorithms and architectures were performed.

The outlook of this project relies on some industrial considerations. Car manufacturers are planning to reduce the cables inside the car and to use a single cable to distribute the main signals multiplexed all around the vehicle. This transition to digital audio signal transmission will foster the use of active loudspeakers, equipped with dedicated digital audio processors, which will be implemented with a DSP, using state-of-the-art tools for simulation, validation and synthesis. The audio processor, designed and tested within this ESD experiment, can be considered a first step in this direction. It will show the feasibility of reducing loudspeaker distortion by digital signal processing, and will exploit digital implementation versatility in order to allow very fast adaptations to different loudspeaker models. Moreover the use of digital technology makes easier the interconnection with those standard protocols that are becoming mandatory in order to respect European laws on EMC and EMI.

Another step will be made in the direction of designing a dedicated hardware, still consistent with the solution proposed in this application experiments, but more attractive from economical and performance point of views. This board could be embedded in the electronic boards now adopted for active loudspeaker systems, and more suitable DSP and CODEC will be adopted, or a dedicated hardware platform could be designed.

The results of the application experiments can be summarized as follows. At first a parametric non-linear model for low frequency loudspeaker was designed. It allows compensating linear and non-linear distortion of the transducer/environment chain, by realizing an audio processor, whose architecture is the inverse of the designed model.

Then equalizer systems, especially fitted for car cockpits, were developed. Specifically an automatic design tool for the synthesis of equalizing audio processors was designed.

It allows obtaining the programming parameters of an efficient audio processor, once the hardware platform is fixed, and starting from standard acoustic measurements of the target car cockpit. Several options are available:

- Design of a "simple" inverse equalizer with FIR or Multi-rate filters.
- Design of an equalizer based on "Warped filters".
- Design of a harmonization system for car cockpits.

All of them were experimentally validated on different cars, relying on the DSP based used, and on several dedicated and commercial software packages.

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