Correction of crossover phase distortion using reversed time all-pass IIR filter

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ABSTRACT

The purpose of this paper is to describe a correction implementation of group delay distortion arising from a two-way loudspeaker system crossover. Having determined an IIR all-pass filter having a group delay response corresponding to that of the system crossover to be corrected, we have validated under “Matlab” and implemented in DSP the time reversal solution proposed by [1] and [2], enabling an IIR filter to be inversed, whilst retaining stability and causality. In addition to theory and calculation validation, we have also carried out preliminary listening tests, supporting the evaluation of timber modification, sound clarity and space localization due to the group delay distortion correction.

1. INTRODUCTION

Within the framework of a project named “Leonardo”, we chose to establish whether sound is likely to be affected by phase distortion. A theoretical study of phase and group delay distortions and appropriate listening experiments demonstrated justifiably that these effects have not to be neglected. It was then decided to analyse the phase and group delay distortions produced by one of our two-way loudspeaker systems, with the aim of providing a solution to correct them.

A two-way vented loudspeaker system comprises three main phase and group delay distortion sources:

- the high-pass filter resulting from the medium and port alignment,
- the low-pass filter resulting from the tweeter high cut-off frequency,
- and the internal crossover.

An example of signal modification due to phase distortion coming from high-pass and low-pass filters is shown in the figure below. The latter represents the band-pass filter time response of a half-squared step signal comprised in the audio frequency range (20Hz-20kHz).

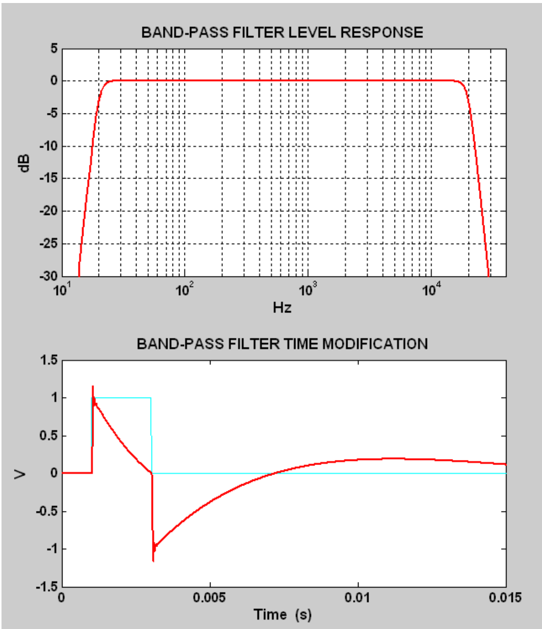

Figure 1 Time modification due to band-pass filter phase distortions

As we can see, the time signal response to a low frequency high-pass filter shows deflection behaviour when compared to the original signal. The high frequency low-pass filter effects are represented as little glitches occurring at transition states.

Some preliminary listening tests gave the appearance that phase and group delay distortions due to band-pass filters are not as noticeable as those produced by crossovers. This assertion has still to be validated. The purpose of this paper, however, will be to explain our approach to crossover group delay correction, according to the concrete example of one of our passive vented loudspeaker system. It is worth noting that the passive crossover of our system has been calculated to obtain a flat on-axis frequency amplitude response. That means that the crossover compensates for irregularities peculiar to the medium and tweeter locations and frequency responses. It is therefore important to correct only the crossover group delay response without touching its amplitude response.

In order to illustrate the phase and group delay distortion phenomenon, the figure below shows the time modification of a squared signal passing through the crossover system. The frequency response amplitudes of these two signals are of course absolutely identical.

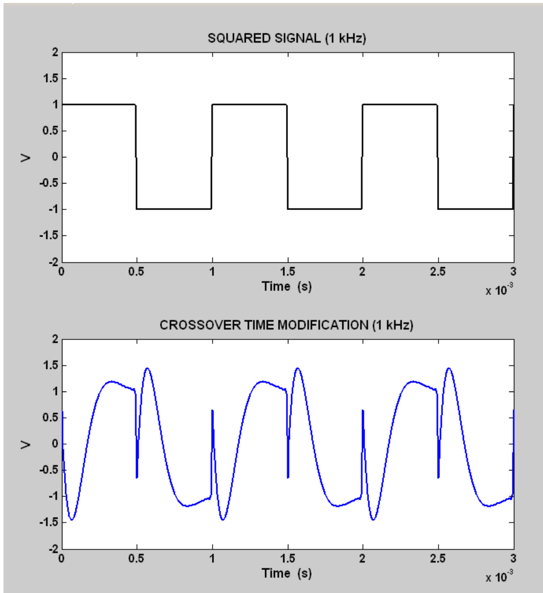

Figure 2 Time modification due to crossover phase distortions

As we can see, the time behaviour modifications due to crossover phase distortions show important problems at transition states, that is an initial rise followed by a dramatic reversal with overshoot before starting the smooth rise. The group delay maximum value of this crossover does not exceed 0.25 ms. This value remains very small compared to the high-pass filter group delay resulting from the medium and port alignment of the vented loudspeaker system, which can reach 10 ms.

The chosen group delay correction method includes in a first step, calculating an IIR all-pass filter with group delay and phase responses as close as possible to those of the crossover to be corrected. In a second step, this all-pass filter is inversed with the aim of obtaining a linear phase response. This inversion is carried out according to the time reversal method, enabling the inversed IIR filter stability and causality to be retained. This method has been validated before being implemented in DSP.

NB: People interested by phase and group delay theoretical study are invited to read the thesis work [3].

2. IIR ALL-PASS FILTER CALCULATION

2.1. Passive crossover calculation

In a preliminary step, the phase and group delay of the loudspeaker system crossover are calculated in the simplest way, that is to say loaded by simple electrical resistances.

The below “Pspice” electrical circuit enables the loudspeaker system crossover transfer function to be calculated according to resistive loads of 8.2 Ohm (see figure below). The resistance values have been chosen to be an accurate reflection of the tweeter and woofer nominal impedance.

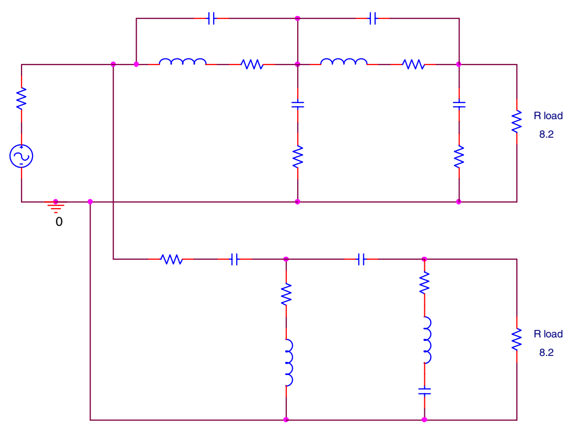

Figure 3 Passive crossover loaded by resistive loads (8.2 Ohm)

2.2. Approximation of crossover phase and group delay

A calculation sheet has been implemented under “Matlab”, enabling the correction IIR all-pass filter coefficients to be calculated in order to approximate the crossover phase and group delay frequency responses. The all-pass filter is determined by its resonance frequency and group delay value at this frequency. These two values are then adjusted in order to find the best approximated phase and group delay frequency responses.

The figure below shows an example of this calculation sheet, where the time representation is given by an impulse signal and a 1 kHz continuous squared signal.

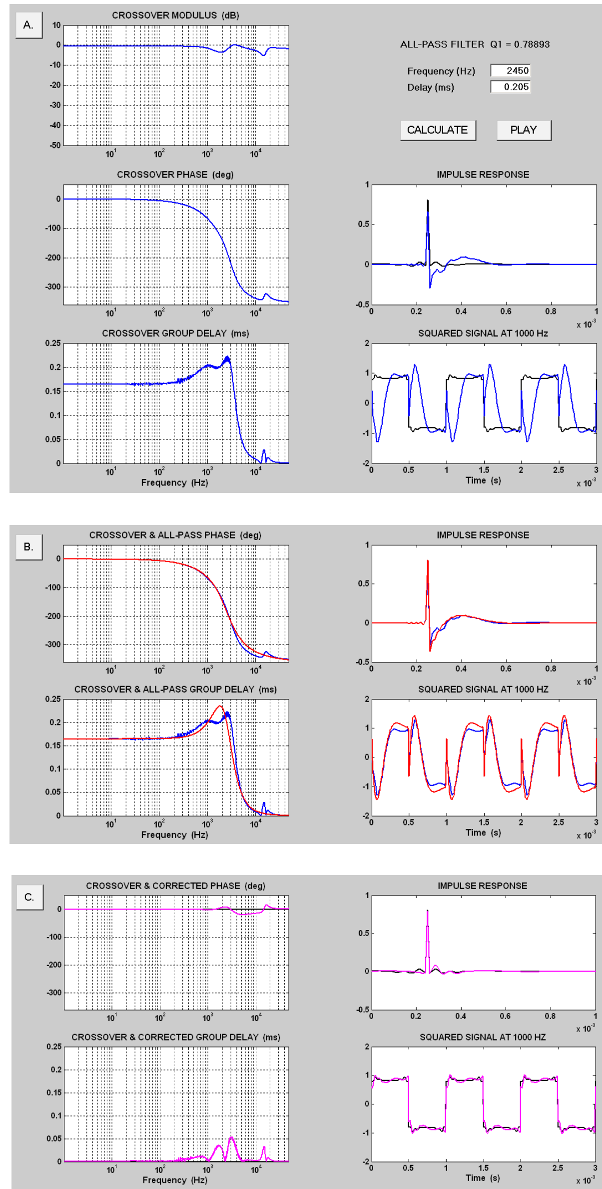

Figure 4 IIR all-pass filter calculation sheet

The figure 4.A shows the simulated crossover frequency response in amplitude, phase and group delay. It shows also the crossover time response superimposed on the desired zero-phase crossover time response.

The figure 4.B shows the crossover response in frequency and time, compared to the responses of the calculated IIR all-pass filter. The latter is chosen in order to approximate the phase and group delay responses of the crossover.

The figure 4.C shows the desired zero-phase crossover frequency and time responses compared to the responses resulting of the group delay correction. The latter comes from the multiplication of the loudspeaker crossover by the inversed of the determined IIR all-pass filter (coefficient inversion).

The calculation sheet enables also the different continuous square signals to be heard.

2.3. Calculation refinement

The modelling of the loudspeaker system crossover has now to be refined in order to calculate the crossover phase and group delay in a more realistic way, that is to say taking into account the real electrical, mechanical and acoustical loads corresponding to the vented loudspeaker system.

The filtered electrodynamic medium and tweeter can be described by lumped-constant circuits comprising electrical, mechanical and acoustic elements, as shown in the figure below. It is not the purpose here to dwell on these well-known parameters [4].

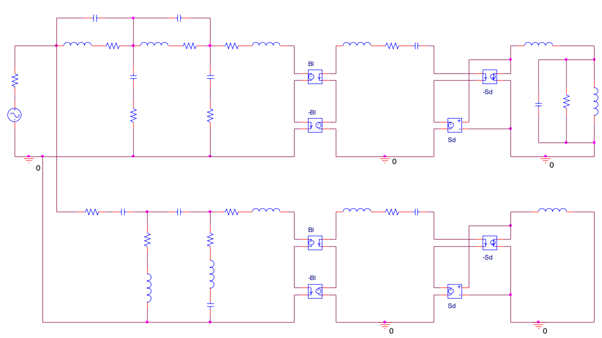

Figure 5 Passive crossover loaded by loudspeaker system load

The all-pass IIR filter coefficients are then readjusted according to the crossover transfer function calculated from the equivalent electrical circuit illustrated above.

In order to validate this loudspeaker system modelling, the measured crossover transfer function has been also introduced in the calculation sheet.

2.4. All-pass IIR filter inversion

As explained above, the correction of the crossover phase and group delay distortions is carried out by multiplying its transfer function by the inverse of the chosen all-pass filter. In the case of IIR filter, the transfer function and its inverse may be written as:

$$H(z) = \frac{\sum_{m=0}^{M} b_m z^{-m}}{\sum_{n=0}^{N} a_n z^{-n}} \qquad H^{-1}(z) = \frac{\sum_{n=0}^{N} a_n z^{-n}}{\sum_{m=0}^{M} b_m z^{-m}}$$

The inverse filter cannot be implemented as it is, due to its non causal (negative group delay) and unstable (pole outside unity circle) behaviour. If it is possible to render it causal by introducing a pure delay before its transfer function, it remains impossible to render it stable without touching its amplitude and degrading its phase response.

The solution to this significant problem is provided by the time reversal method.

3. TIME REVERSAL METHOD

3.1. Procedure and validation

This inversion method, proposed by [1] and [2], offers the advantage of calculating the IIR filter H(z) instead of its non-causal and unstable inverse. The inversion is then carried out by inversing the samples before and after the filter H(z). The complete method has been simulated in order to validate the theory with the view of DSP implementation. The figure below shows the block diagram of the simulated implementation.

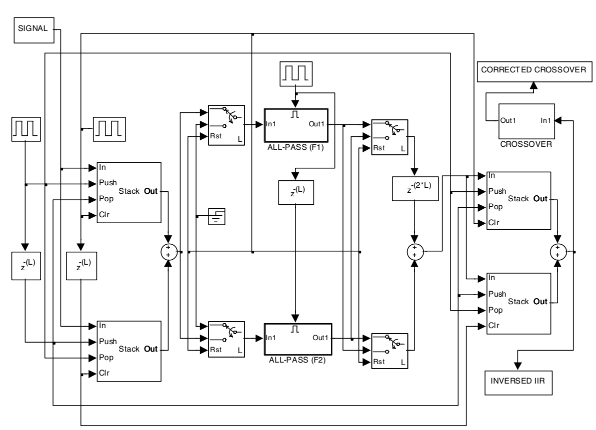

Figure 6 IIR all-pass filter inversion (block diagram implementation)

The input signal is acquired in consecutive sections of the length L samples. The sections are time reversed (input LIFO buffers) and separated on two signals to be taken half the time and completed by zeros to form sections of the length 2L samples. Both signals are filtered by identical causal and stable IIR all-pass filters, which are initialised at the start of each double section. Both signals are then recombined in leading and trailing sections. The leading sections signal is delayed by 2L samples before being added to the trailing sections. Finally, the sections are again time reversed (output LIFO buffers). The IIR all-pass filter coefficients correspond to the last chapter calculated all-pass IIR filter.

The figure below shows the result of this method for a half-squared step input signal (first curve). The second curve corresponds to the IIR all-pass filter and the third curve to the time-reversed all-pass IIR filter obtained by time reversal method. The 4L delay comes from the necessary LIFO buffers. The validation is given by the last curve, resulting to the cascade of the IIR all-pass filter (2nd curve) and the time-reversed IIR all-pass filter (3rd curve).

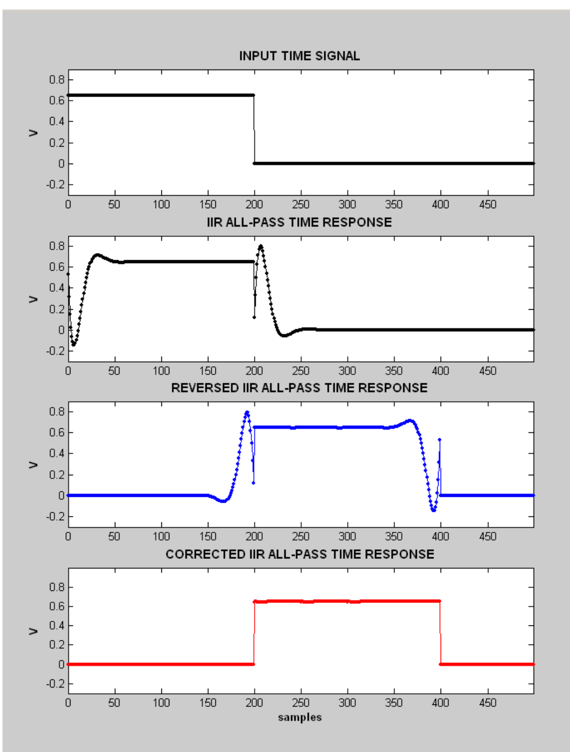

Figure 7 Block diagram results

As we can see, the result shows a high degree of accuracy compared to the input signal. The small irregularities come from the fact that the signal has been cut in L samples sections, meaning that the Infinite Impulse Response has been calculated into finite sections. The purpose is then to find a good compromise between section length wide enough to minimize the irregularities and small enough to be calculated in DSP memory.

3.2. DSP implementation

In order to carry out a listening test of crossover group delay correction, the time reversal method has been then programmed in DSP, following the block diagram represented in the figure below, where each little square represents a section of length L samples.

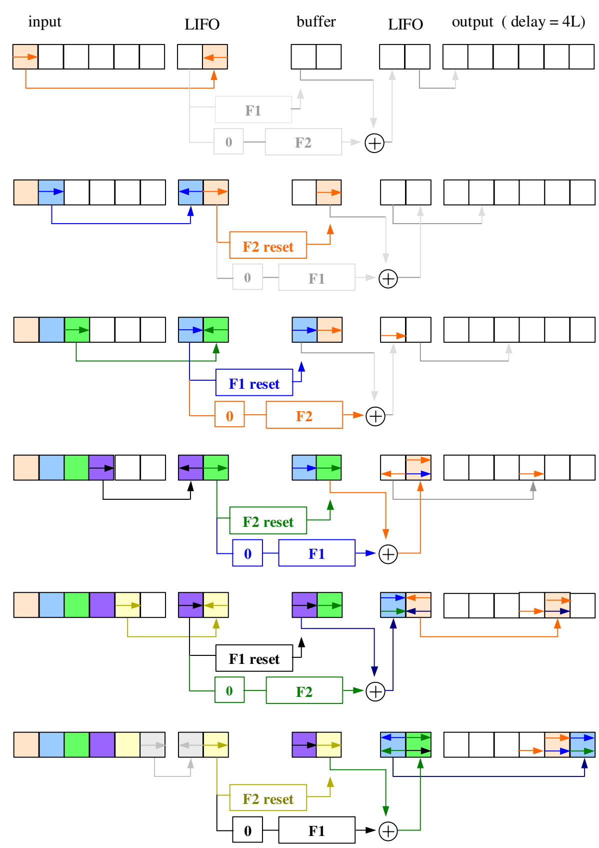

Figure 8 DSP implementation

4. PERCEPTION OF CROSSOVER PHASE AND GROUP DELAY CORRECTIONS

A listening test has been carried out in order to understand the phenomenon of phase and group delay distortions. The test simply comprised listening to a music track 2 minutes long, first without correction; then a second time with crossover group delay and phase corrections.

The results were extremely impressive for all listeners. The majority of them have noted that the corrected track presented greater sound clarity, a more spacious and detailed sound environment, as well as best stage localisation.

In the future, we will plan more serious psychoacoustic tests in order to validate the above perceptions related to crossover group delay and phase distortions.

5. CONCLUSION

This paper concludes with success the first stage of our “Leonardo” project. Based on these first very promising results, we have decided now to correct the whole loudspeaker system phase and group delay distortions. To do that and according to the restrictive calculation of IIR filter approximation method, we have decided to use FPGA instead of DSP in order to implement FFT-IFFT calculation, enabling the phase and group delay corrections to be carried out in the frequency domain. This method enables band-pass, high-pass and low-pass filters phase and group delay distortions to be easily corrected.

6. REFERENCES

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