Implementation of Specific Equalising Filters using Field Programmable Analogue Arrays (FPAA)

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Abstract – Following a short description of Field Programmable Analogue Arrays (FPAA), the implementation of specific equalising filters, using FPAA, will be presented. Special focus will be laid on the design and implementation of equalising filters for digital to analogue conversion, but the design program can solve a by far wider range of filter implementation problems. Included in the paper is an introduction to FPAA, a description of the design process and a presentation of some results.

I. INTRODUCTION

The idea of using FPAA for the variable design of analogue circuits dates back to the development of the first integrated circuits. But, as the progress in programmable digital circuits rushed forward in huge steps, the evolution of programmable analogue circuits proceeded only slowly. Solely in the field of programmable analogue filters a real breakthrough appeared. Meanwhile, however, a real boom seems to happen. Configurable analogue circuits, based on operational amplifiers, or switched capacitor technique flood the market. Especially dynamically reconfigurable Field Programmable Analogue Arrays allow for analogue signal processing algorithms, formerly restricted to digital signal processing. Adaptive analogue circuits for highly sophisticated purposes, handling signal frequencies in the MHz region with tolerances smaller than 0.1%, can replace digital signal processors.

In the paper, the design and implementation of an equalising filter, using the anadigm FPAA AN221E04, is presented. After a short outline of the design method, the FPAA’s hardware constraints are demonstrated. Measured results show the capability of the design program and the FPAA circuit. The test for not to surpass constraints, embedded in the design software, helps preventing the user from unexpected surprises.

II. PROGRAMMABLE ANALOGUE DEVICES

Field Programmable Analogue Arrays (FPAA), based on configurable analogue blocks (CAB), consisting of operational amplifiers, passive elements and a programmable connecting network can be combined to highly qualified circuits such as filters, integrators etc. Fig. 1 shows the block circuit of a typical FPAA, the anadigm AN221E04.

Time continuous and time discrete FPAA differ in technique and performance. Traditional analogue arrays depend on integrated resistors and capacitors, and, thus yield only low precision at 5% to 10%. But, time discrete systems, like the AN121E04 and AN221E04 using switched capacitor technique to implement the desired analogue functions, only depend on the ratios of capacitors. Tolerances down to 0.1% are the advantage achieved easily. Fig. 2 shows a typical structure of such a biquad filter in sc-technique.
amplitude at the resonance frequency, known from filters with operational amplifiers, a shift in the resonance frequency can be measured in the sc-filter device. See Fig. 3.

![Fig. 3. Pole shift due to high pole quality factor](image)

This restriction was included in the design program, avoiding those differences in the design. [2],[3]

III. POST DAC EQUALISING FILTERS

Analogue post DAC filters for smoothing the stepfunction shall fulfil two different purposes, see Fig. 4.

1. Equalising the non-linear frequency response of the digital to analogue converter \( H_{DA}(j\omega) \), following equation (1).
2. Suppressing the alias spectra due to the sampling process, hence smoothing the signal. Equation (2).

\[
H_{DA}(j\omega) = T \cdot e^{-j\omega T/2} \left( \frac{\sin \omega T/2}{\omega T/2} \right)
\]

\[
H_{C2}(j\omega) = \frac{1}{H_{DA}(j\omega)} \quad |\omega| \leq \omega_g
\]

\[
H_{C2}(j\omega) = 0 \quad |\omega| > \omega_g / 2
\]

A. DESIGNING A STARTING SOLUTION BY MEANS OF INTERPOLATION

Following the common design method, an appropriate characteristic function \( K(j\omega) \) is determined by implementing the requirements for the pass- and the stopband. The denominator places the zeroes for the desired attenuation in the stopband and the numerator is interpolating the desired passband response. Equations (3)-(6).

\[
|H_{C2}(j\omega)|^2 = \frac{b_m^2}{1 + |K(j\omega)|^2} = \frac{b_m^2}{1 + \frac{F(\omega)}{N(\omega)}}
\]

\[
N(\omega) = \prod_{\mu=1}^{m} (\omega^2 + \omega_0^2)^2
\]

\[
F(\omega) = \sum_{\lambda=0}^{n+1} \frac{(\omega - j\omega_\lambda)^2}{(\omega - j\omega_\lambda)^2} \prod_{\nu=0,\nu\neq\lambda}^{m} (\omega^2 - \omega_\nu^2)^2
\]

B. OPTIMISING THE STARTING SOLUTION

In a second step, we optimise the starting solution for best performance due to the error criteria defined in equations (7), (8) and (9). Applying Nelder and Mead’s simplex algorithm already yields an excellent smoothing filter, but results in only a near Chebyshev behaviour in the passband. Solving the non-linear equation system (10) using Newton Raphson optimisation then leads to optimum DAC filters. Fig. 5.

\[
\varepsilon = k \cdot \max \{ |E(j\omega)| \} + \varepsilon_R \Rightarrow \min
\]

\[
E(j\omega) = 1 - |H_{DA}(j\omega)H_{C2}(j\omega)| / T \quad |\omega| \leq \omega_c
\]

\[
\varepsilon_R = \int_{-\omega_c}^{\omega_c} |H_{DA}(j\omega)H_{C2}(j\omega)|^2 d\omega
\]
Fig. 5. Optimised post DAC filter, starting solution (left) and result (right)
IV. IMPLEMENTATION OF POST DAC FILTERS ON FPAA

Following the design process, outlined in III, the transfer function of filter $G_{54_2.35}$ of Fig. 5. was calculated in [1]. Then, obeying the restrictions of the FPAA hardware it was implemented on AN221E04 using the software developed in [2].

![Graph showing amplitude frequency response](image)

**Fig. 6. Amplitude frequency response of the G 54_2.35 filter, (calculated and measured)**

The results, both determined and measured are compared in Fig. 6. An excellent approximation of the desired frequency response holds for nearly the whole passband. Only next to the cutoff frequency, an increase in deviation of up to approximately 2%, compared to the theoretical solution, arises. Fig. 7.

![Graph showing deviation of amplitude](image)

**Fig. 7. Deviation of amplitude in the passband**

To show the overall performance of the cascaded DAC and the filter, Fig. 8 shows the frequency responses of the DAC, the designed filter and the product of them.

![Overall performance of DAC and post DAC filter](image)

**Fig. 8. Overall performance of DAC and post DAC filter**

V. CONCLUSION

As was shown, the proposed design method, combined with an a priori taking into account of the FPAA’s hardware restrictions yield an equalising filter meeting the given prescriptions as closely as desired. Excellent equal ripple behaviour in the passband and high stopband attenuation are achieved in the result. Though, the proposed filter was only an example showing the capability of the filter design program of [2] and the AN221E04 FPAA hardware, an easy and qualified filter implementation can be expected, even in the case of high demands.

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VII. REFERENCES


