Irina DORNEAN, Eng.

Abstract of PhD Thesis


PH.D. COMITEE COMPONENCE:

COORDINATOR:  -Prof.dr.eng. Gabriel OLTEAN – Electronics, Telecommunications and Information Technology Faculty, Technical University of Cluj-Napoca;

SCIENTIFIC COORDINATOR:  -Prof.dr.eng. Marina Dana ŢOPA – Electronics, Telecommunications and Information Technology Faculty, Technical University of Cluj-Napoca;

MEMBERS:  -Prof.dr.eng. Alexandru ISAR – Electronics and Telecommunications Faculty, “Politehnica” University of Timisoara
-Prof.dr.eng. Florin SANDU - Electrical Engineering and Computer Science Faculty,"Transilvania" University of Brasov
- Prof.dr.eng. Sorin HINTEA - Electronics, Telecommunications and Information Technology Faculty, Technical University of Cluj-Napoca
Chapter 1  .................................................................................................................................
1. Introduction .........................................................................................................................
  1.1. Motivation ....................................................................................................................... 
1.2. State of the Art .................................................................................................................... 
1.3. FPGA vs. DSP ..................................................................................................................... 
1.4. Multirate Digital Signal Processing..................................................................................... 
1.5. Optimized Algorithm Designs............................................................................................ 
1.6. Objectives and Thesis Organization..................................................................................... 

Chapter 2  .................................................................................................................................
2. ADAPTIVE FILTERS ............................................................................................................. 
  2.1. Introduction ....................................................................................................................... 
  2.2. Applications of Adaptive Filters ......................................................................................... 
    2.2.1. System Identification .................................................................................................... 
    2.2.2. Channel Equalization .................................................................................................. 
    2.2.3. Echo Cancellation for Speech-Band Data Transmission ............................................ 
    2.2.4. Signal Prediction ......................................................................................................... 
  2.3. FIR Filter Structures .......................................................................................................... 
    2.3.1. Transversal Structure .................................................................................................. 
    2.3.2. Symmetric Transversal Structure ................................................................................. 
    2.3.3. Lattice Structure ......................................................................................................... 
  2.4. Adaptive Filter Algorithms ............................................................................................... 
    2.4.1. Least Mean Square Algorithm ...................................................................................... 
    2.4.2. Normalized LMS Algorithm ....................................................................................... 
    2.4.3. Sign Algorithms ......................................................................................................... 
    2.4.4. The RLS Adaptation Algorithm ................................................................................... 
  2.5. Experimental Results and Contributions .......................................................................... 
    2.5.1. Noise Cancellation Application .................................................................................... 
    2.5.2. System Identification Application .............................................................................. 
    2.5.3. Signal Prediction Application ....................................................................................... 

Chapter 3  .................................................................................................................................
3. Digital implementations .......................................................................................................... 
  3.1. Introduction ....................................................................................................................... 
  3.2. Hardware Description Languages ...................................................................................... 
  3.3. Xilinx DSP Solutions .......................................................................................................... 
    3.3.1. Nexys Circuit Board ..................................................................................................... 
    3.3.2. ML501-Virtex 5 .......................................................................................................... 
  3.4. Experimental Results and Contributions to FPGAs Implementations ............................. 
    3.4.1. Hardware Implementation of the LMS Adaptive Algorithm ....................................... 
    3.4.2. Hardware Implementation of the NLMS Adaptive Filter ........................................... 
    3.4.3. Reverberators ............................................................................................................. 
      3.4.3.1. Schroeder Reverberator – Verilog Implementation ................................................ 
      3.4.3.2. Schroeder Reverberator (II) .................................................................................. 
      3.4.3.3. Moorer Reverberator ............................................................................................. 
      3.4.3.4. Gardner Reverberators ..........................................................................................
3.4.3.5. Dattorro Reverberator .................................................................

3.5. Conclusions ....................................................................................

Chapter 4 ..............................................................

4. Multirate digital signal processing .................................................

4.1. Introduction .................................................................

4.2. Design of FIR Filters Using Window Functions ......................

4.2.1. Kaiser Window ..............................................................

4.2.2. Blackman Window .........................................................

4.2.3. Bartlett Window .............................................................

4.2.4. Hamming Window...........................................................

4.3. Decimation and Interpolation .....................................................

4.4. Filter Banks- Analysis and Synthesis ...........................................

4.5. Subband Coding ........................................................................

4.6. Experimental Results and Contribution ....................................

4.6.1. MATLAB/Simulink Subband System Identification Application

4.6.2. Optimized Algorithm Designs ................................................

4.6.2.1. Fullband Design System Identification Application

4.6.2.2. Two Subbands Coding Design of System Identification Application

4.6.2.3. Four Subbands Coding Design of System Identification Application

4.7. Conclusions ....................................................................................

Chapter 5 ..............................................................

5. Conclusions and Contributions .......................................................
**Motivation**

Signal processing technique is an important instrument regarding various fields in science and engineering. Commonly, the characteristics of systems are either not known or time variable due to many reasons, undesirable in many cases. Therefore, processing approaches should adapt to the unknown characteristics, hence to extract valid information in a changing scenario. For that reason the adaptive algorithms should be simple, computationally efficient, implementable on the existing hardware platform (like digital signal processors or configurable blocks) and cost effective for commercial utility. Any real time processing has to be adaptive and some of the common applications are: compression and coding, active control of noise (inside aircraft cabin and automobiles, industrial noise) and vibration, communication applications like channel equalization, acoustic and line echo cancellation, adaptive antenna arrays and adaptive processing of biomedical signals.

In order to implement economically these applications using modern tools and structures, multirate algorithms are the prime choice, due to their high computational efficiency. Typical structures of multirate systems are models for the design of efficient architectures for microelectronic circuits. The increased evolution of multirate signal processing is balanced by the emergence of new applications, like sub-band coding of speech, multicarrier data transmission, fast transforms using digital filter banks and wavelet analysis of all types of signals.

The trend in the electronics market is to shift the FPGA-based architecture for both electronic devices and test instrumentation. This type of programmable structures have drastically increased in logic cell counts and functionality and decreased in cost per transistor. Vendors are also beginning to integrate FPGAs with devices such as processors, data converters, and transceivers to deliver increased performance and user programmability even closer to the I/O pin. FPGAs were usually programmed using hardware description languages (HDLs) such as VHDL and Verilog, and many designers implementing signal processing functions into configurable logic today still prefer this method. Programming techniques have advanced and include design entry from platform such as MATLAB and Simulink, which are more common to non-traditional FPGA designers such as system engineers and DSP algorithm developers and programmers.

The vision of Xilinx is that multiple tools can interface and allow today's diverse groups to essentially work in the language of the problem with which they are most familiar. The diagram shows that today, FPGA-based signal processing systems can be built using a combination of Simulink models, MATLAB code, HDL code and C. With such design tools, designers can manipulate the FPGA's resources and even program the FPGA from high-level design tools like MATLAB/Simulink and MATLAB.
Conclusions and Contributions

The thesis approaches the field of adaptive signal processing, following step by step new methods of development of the specific applications. The measure of performance is represented by the cost optimization of design and implementation of the system. Gradually, are attended:

- Performance detection of the adaptive algorithms;
- Digital implementation improvement of the adaptive filters and several types of reverberators;
- Execution time optimization of the adaptive algorithms by multirate systems processing procedure;
- Designing cost minimization by using automated code generation;
- Verification of the implemented application by modern testing tools.

General aspects of the adaptive filters design are studied, by the software design of its principal applications, such as system identification, channel equalization, echo cancellation for speech-band data transmission and signal prediction. An overview over the structures of digital FIR filters (transversal, symmetric transversal and lattice structures) is made.

The adaptive filter algorithms are described in detail: the least mean square method that makes successive corrections to the weight vector and leads to the minimum mean square error; the normalized LMS algorithm, commonly used in applications due to its fast convergence and stability; the sign algorithms that attempt to simplify the computational requirements of the LMS by reducing the number of multiplications required; the recursive least squares algorithm which uses information from all past input samples to estimate the autocorrelation matrix of the input vector.

In order to determine the performances of these kinds of filters the author accomplished the stages:

- Matlab/Simulink designs were developed for suggestive applications of the adaptive algorithms: noise cancellation, system identification and signal prediction.
  - Four adaptive algorithms were tested in parallel with the same input and confirmed the NLMS algorithm to be the fastest and most stable from all: Least Mean Square, Normalized Least Mean Square, Sign-Error Least Mean Square and Sign-Sign Least Mean Square. The MATLAB/Simulink environment was chosen because its calculus precision and complexity; another reason was that it handles very easy the floating point operations.
Statistics made concerning the performance of the adaptive algorithms. The proof of NLMS superiority was computed in terms of weights approximation and error convergence. It was tested in different input conditions, parallel compared with other types of adaptive filters.

Due to the fact that configurable structures become very popular, arises the need of describing the circuit’s operation, its design and organization, and tests to verify its operation by means of simulation through hardware description languages. Adaptive filter structures (LMS and NLMS) and reverberators (Schroeder, Gardner, Moorer and Dattorro) implementations using HDL coding are detailed presented in the thesis. Comparison between simulated digital algorithms and Simulink developed models was done.

The system identification application was in detail studied by the author, following the steps:

- The first step was to create a Simulink model which provided the input signals for the hardware design also, in order to compare the results for the two implementations. From the Simulink results it is obvious that as the error converges to a minimum value, the output signal takes the same values as the desired signal.

Concerning the hardware implementation:

- The LMS adaptive filter core was coded in Verilog-HDL, a dedicated description language. Simultaneously, a detailed test-bench was written to verify the LMS adaptive filter core. All the test files and the filter core were instantiated into a top level file to be simulated synchronously. The results prove the same thing as in the Matlab design, that in time, the coefficients adapt and the error converges. The results were tested on the Digilent’s Nexys circuit board, an integrated circuit development platform based on a Xilinx Spartan 3 FPGA.

- Also from the hardware point of view, the NLMS algorithm was studied. A HDL implementation of system identification application having a NLMS adaptive filter was implemented. Each individual component was created using behavioral architecture of the Verilog hardware description language. The input signals (noise) were provided from MATLAB®/Simulink®, where the unknown system is modeled. These signals are transformed into binary data files which characterize them at each clock cycle. In order to verify the implementation, the test files together with the design file were compiled and simulated using ModelSim environment. Plots regarding the convergence of both NLMS and LMS algorithms were provided using a special developed module that transforms data from ModelSim into MATLAB® plots.
Concluding the experimental results, the statistics made in the Simulink applications are confirmed. The NLMS algorithm is better than the conventional LMS adaptive algorithm, because it is more stable and less noisy; also the convergence speed is considerably increased. It is feasible to implement variable step size based adaptive filters in the digital domain.

**Contributions** are measured in the code optimization by the improvement of the mean squared error convergence speed. For a better precision, floating point operations should be performed in the hardware implementation. This code can be the source of programming a large area of FPGA devices because modern chips contain many resources that support DSP applications, which are optimized for high performance and low power consumption.

Acoustic environment of the unknown system identification application can be modeled by artificial reverberators. Reverberators are key elements in three-dimensional spatial audio reproduction. The richness and spaciousness of reproduced sound can be enhanced with these structures. An inadequate choice of the filter parameters often results in audible artifacts such as metallic and ringing sound.

The author **contributed** by

- Verilog digital implementation of four types of artificial reverberators (Schroeder, Gardner, Moorer and Dattorro), each having its specific versions for different categories of applications. The acoustical parameters were studied and the reverberation phenomena principles were considered. Description of the blocks in MATLAB®/Simulink® environment was made, the obtained results being considered as a reference for the next step. The main advantage of the digital implementation of these types of reverberators is that they can be used for acoustic signal processing and downloaded on development boards having FPGA components. The field programmable gate arrays offer higher processing speed than the common methods involving digital signal processors.

Execution time optimization of the adaptive filters application of system identification in an acoustical environment is aimed by approaching multirate digital signal processing. In complex adaptive applications, filter structures with high orders suffer from decreased speed convergence and high computational complexity. Subband adaptive filtering technique seems to be able to eliminate these disadvantages. Modern tools were used in order to design high-performance systems using FPGAs.

The **author** studied the behavior of multirate systems in adaptive system identification application by considering the steps:

- A number of Simulink models were designed, having different types of FIR filters, categories of subbands (2, 4 and 8) and values for the adaptive algorithm
parameters. The input information is represented by a speech. All data is presented in tables and statistic plots.

- The example of system identification application using 4\textsuperscript{th} order Blackman FIR window filter for analysis and synthesis and a four band decomposition of the input signal was selected for detailed study. The normalized version of the Least Mean Square algorithm was used, due to its robustness and simplicity. As expected, after the multirate processing, the system’s behavior changes according to the number of decomposed subbands. The reason is the reduced number of computations on each subband. Also it appears the unwanted alias effect, minimizing the quality of the signal.

ERLE defines the measure of performance, as the ratio of the power of the desired signal over the power of the residual signal.

The example of system identification application having a subband decomposition of the input signal was implemented also using a special concept, \textit{Xilinx System Generator}. It is an extension of Simulink environment from Xilinx and consists of a Simulink library called the Xilinx Blockset that can be mapped directly into target FPGA hardware. The utilized target board was Digilent Spartan-3E, for its specific digital signal processing facilities.

Being a new concept, \textbf{contributions} in developing complex systems are brought:

- For the application of adaptive system identification, three structures were proposed:
  - a full band system,
  - a system by two subbands decomposition of the input signal,
  - a system of four subbands decomposition.

  The low-pass and high-pass FIR filters were designed by window functions, the determination of analysis and synthesis filter’s coefficients being an important task in alias elimination introduced by the multirate systems. Simulations were done considering the specific signals, which confirm the performance of the implemented systems.

- Concerning the verification step, hardware co-simulation was performed. The results prove the behavior of the designed system in terms of error convergence through weights adjustment.

- Estimation of the needed FPGA resources is done for each case. XSG is an instrument that provides information for designers having different skills allowing the development of complex applications.