An Efficient FPGA Simulink Design Based DCT Transform Architecture for Signal Denoise Application

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Abstract

The communication industry field is mainly focused by high data transfer and more channel capacity in mobile communication. VLSI technology is used to modify any type digital based hardware architecture and to reduce the hardware system power, speed and complexity level. The filter process is mainly used to DSP and DIP real world application. The filter process is to remove the noise in original signal or image. So the filter architecture optimized process, to reduce the filter processing time and to increase the performance. Adaptive digital filters find wide application in several digital signal processing (DSP) areas. FIR filter architecture is used to effectively remove the noise in received channel data bits and to require less circuit complexity.The existing system frequency transformation based filters (FT filters) provide an absolute control over the cutoff frequency. However, the cutoff frequency range of the FT filters is limited. The second-order frequency transformations combined with coefficient decimation technique based filter (FTCDM filter) has wider range compared with the FT filter. But this method have some limitation. So we use the discrete cosine transform (DCT) modulation based low pass filter transformation process. Proposed system is to design a 18-band DCT transform signal denoising architecture simulink design. This simulink design is to consist the XILINX based digital architecture block. Discrete cosine transform (DCT) modulation are utilized to generate a uniform 18-band filter bank first, and then all elements of are replaced by all-pass filters to obtain a non-uniform filter bank. A fast recursive structure and variable-length algorithm is further developed to efficiently accomplish DCT modulation. Proposed system is to improve the output signal reconstruction effectively. Proposed system is to reduce the latency level and to increase the application system speed level.

Keywords: Filter Processing Time, FTCDM Filter, XILINX, Discrete Cosine Transform, Latency Level

I. INTRODUCTION

A Discrete Cosine Transform (DCT) expresses a finite sequence of data points in terms of a sum of cosine functions oscillating at different frequencies. DCTs are important to numerous applications in science and engineering, from lossy compression of audio (e.g. MP3) and images (e.g. JPEG) (where small high-frequency components can be discarded), to spectral methods for the numerical solution of partial differential equations. The use of cosine rather than sine functions is critical for compression, since it turns out (as described below) that fewer cosine functions are needed to approximate a typical signal, whereas for differential equations the cosines express a particular choice of boundary condition.In particular, a DCT is a Fourier-related transform similar to the discrete Fourier transform (DFT), but using only real numbers. DCTs are equivalent to DFTs of roughly twice the length, operating on real data with even symmetry (since the Fourier transform of a real and even function is real and even), where in some variants the input and/or output data are shifted by half a sample. There are eight standard DCT variants, of which four are common.

The most common variant of discrete cosine transform is the type-II DCT, which is often called simply "the DCT" its inverse, the type-III DCT, is correspondingly often called simply "the inverse DCT" or "the IDCT". Two related transforms are the discrete sine transform which is equivalent to a DFT of real and odd functions, and the modified discrete cosine transform (MDCT), which is based on a DCT of overlapping data.

II. ADDITIVE WHITE GAUSSIAN NOISE (AWGN)

AWGN is often used as a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude. The model does not account for fading, frequency selectivity, interference, nonlinearity or dispersion. However, it produces simple and tractable mathematical models which are useful for gaining insight into the underlying behavior of a system before these other phenomena are considered.

Many studies presented various effective filter bank designs with different structures for digital hearing aids. Generally speaking, it is divided into two strategies: One is uniform filter bank and the other is non-uniform filter bank. Considering the nature of human's auditory, the technique of dividing the frequency non-uniformly is much suitable for digital hearing aids. Thus, many non-uniform filter bank approaches are recently devoted to satisfy the needs of the hearing impaired with patient's audiogram result. Nielsen and Sparso proposed seven-band IFIR filter bank with asynchronous circuit techniques and low power scheme.

III. EXISITNG SYSTEM

Visual communication is an important aspect of music performance, for example, to pick up temporal cues and find the right entries. Visual cues can also be instrumental to negotiate the solo order in improvised music or enable social exchange, for example, by signaling someone that her solo was well received.

The problem with visual communication is that one has to catch someone else's attention, and visual cues outside someone's visual field cannot be detected, even more so if the addressee is busy reading a music score or closing his eyes in a Free Music session. Acoustic communication does not encounter these challenges, but of course someone does not want to disturb the music with other acoustic signals. The haptic modality has the advantage that it does not necessarily interfere with the acoustic signal and does not require attention. However, it allows interpersonal communication if both parties are within close proximity. Using telematic interfaces solves the problem of proximity by allowing participants to communicate over any physical distance.

The double bass and cello sections in the orchestra transmit vibrations to the stage floor through the end pins. Whether or not these vibrations may contribute to the perceived sound in the all has been investigated since the 1930s. In this study, the conditions for an efficient transfer of instrument vibrations to the floor, as well as the radiation from the floor to the audience area, are investigated. The study includes measurements of the impedance matching between bass and stage floor, the vibration velocity transfer to the floor via the endpin, and radiation from point-driven bending waves in the stage floor well below the coincidence frequency.

The impedance conditions and radiation properties for the stage floors of five concert halls were investigated. In the most promising hall, a full-scale experiment was run with an artificially excited double bass supported via the end pin on the stage floor, and on a concrete support below, respectively.

The coupled perception of sound and vibration is a well-known phenomenon during live pop or organ concerts. However, even during a symphonic concert in a classical hall, sound can excite perceivable vibrations at the body surface. However, the concert visitor might not be aware of those vibrations, because the tactile percept is integrated with the other senses into one multi-modal percept. This article discusses the influence of whole body vibrations on the listener experience during the reproduction of concerts recordings. Four sequences were selected from classical and modern music, which include low frequency content (e.g., organ, kettledrum, contrabass). A stimulus length of 1.5 min was chosen in order to provide enough time for habituation. The audio signal was reproduced using a surround setup. Additional seat vibrations have been generated from the audio signal. Test participants were asked to rate the overall quality of the concert experience. The results show that vibrations have a significant influence on our perception of music. This finding is interesting in the context of audio reproduction, but also for the construction of concert venues.

In spite of its importance for the understanding of the evolution of sound communication, information concerning the vocal world of crocodilians is limited. Experimental works have brought evidence of the biological roles of juvenile sound signals, with "hatching calls" eliciting care by the mother and synchronizing clutch hatching, "contact calls" gathering groups of juveniles, and "distress calls" inducing maternal protection. Recently, we investigated the question of species-specific information coding within juvenile calls. The analysis of signal acoustic structure shows inter-specific differences between calls. However, using playback experiments, we bring the evidence that these differences are not relevant to animals, either juveniles or adults. By using calls modified in the temporal and the frequency domains, we isolate the acoustic cues necessary to elicit a behavioral response from receivers, underlying the importance of the frequency modulation slope. Considering previous results underlying the absence of information about individual identity in juvenile calls, we make the hypothesis that these signals basically support a "crocodilian" identity.

IV. DISADVANTAGES

- It consumes more power and to increase the circuit complexity level.
- To affect the system performance level.
- To consume more time due to the write operation

V. PROPOSED SYSTEM

Conductive hearing loss means the sound is not conducted well through a disordered outer or middle ear. Sensor neural hearing loss (SNHL) means the sensory cells in the cochlea are absent or not functioning appropriately. If both conductive and sensorineural losses are present, the result is mixed hearing loss. Conductive hearing loss can be recovered after some adequate treatments, but most people with SNHL are fitted with hearing aids. SNHL can degrade the functions of human ear in several different ways and introduce phenomena such as a raised hearing threshold, decreased and squeezed hearing range, reduced temporal and spectral resolution, and the loss of noise tolerance.

The proposed system is used to denoise the image using the simulink with XILINX blocks based design. This type of architecture is to combined the two domains. The proposed work is to modify the regular filtering process in input noisy speech signal and to analysis the internal high pass and low pass filtering samples. The proposed speech signal filter architecture work is to convert the simulink design into hardware code. And to optimize the filtering processing time.

This objective lead to the use of Xilinx System Generator, a tool with a high- level graphical interface under the Matlab, Simulink based blocks which makes it very easy to handle with respect to other software for hardware description. The various applications where image filtering operations applied are noise removal, enhancing edges and contours, blurring and so on. This paper presents an architecture of filtering images for edge detection using System Generator, which is an extension of Simulink and consists of a models called "XILINX BLOCKS", which are mapped into architectures, entities, signs, ports and attributes, which Scripts file to produce synthesis in FPGAs, HDL simulation and developments tools. The tool retains the hierarchy of Simulink when it is converted into VHDL/Verilog. There are many research works related to image processing and itsreal time implementation using XSG which uses high end hardware similar to the one used in paper by Sami Hasan, Alex Yakovlev and Said Boussakta et al, complicated design used in paper by Zhang Shan shan et al, but the proposed design in this work eliminates the design complexity, takes least resource usage and also implemented in low cost basic FPGA device.

The small co-efficient are dominated by noise, while coefficient with large absolu te value carry more signal information than noise. Replacing noise co-efficient small coefficients below a certain threshold value by zero and an inverse wavelet transform may lead to a reconstruction that has lesser noise. Xilinx system generator provides a simpler and useful tool for developing computer vision based algorithms. It is a more suitable and beneficial option if compared to designing using VHDL or Verilog hardware descriptio n languages (HDLs). The result shows tha t the strength of applying Edge detectionwhich makes it more sensitive in detecting edges in any image and hence more edges are detected using this method.

Audiologists usually identify and diagnose hearing loss with the pure tone audiogram test, which uses sinusoidal signals over octave frequencies from 250 Hz to 8 kHz to measure the minimum levels of sound (i.e., hearing thresholds). The results of PTA test are generally recorded on an audiogram. demonstrates a typical example of moderate-to-severe hearing loss. Fitting hearing aids usually requires a prescription formula. The widely used NAL-NL1 or the HSE for Chinese , generates the ideal electro-acoustic response (i.e., the gain-curve) of a hearing aid. The gain-curve specifies the insertion gain, or the amplification, at each standard 1/3-octave frequency from 150 Hz to 8 kHz. The goal of the NAL-NL1 is to maximize the speech intelligibility while maintaining the loudness of the amplified sound equal to, or less than, that normal hearing. The NAL-NL1 produces different gain-curves for different input sound pressure levels.

Advanced hearing aids are currently battery-powered digital devices consisting of a microphone. The microphone and the receiver perform the transformation between a coustic and electrical signals. The DSP circuit performs sophisticated functions including the auditory compensation algorithm to overcome the hearing loss, and noise reduction and feedback cancellation to improve speech quality and intelligibility. The DSP circuit also uses adaptive directional microphones and spectral shaping for speech enhancement.

According to Kates, a DSP block, performing entire set of DSP functions, typically consumes up to 61% of the overall power budget of a digital hearing aid. One common approach to realize the auditory compensation algorithm, which makes the sound audible for hearing-aid wearer, is to employ an analysis filter bank followed by sub-band amplification and multi-channel wide dynamic-range control andansynthesisfilter bank , A low power Mandarin-specific hearing aid test chip was recently implemented in UMC 90 nm CMOS technology with High-VT standard cells. The test chip contains an 18-bandfilter bank and 3-channel WDRC auditory compensator and a multi-band noise reduction with entropy enhanced voice activity detection (VAD). The power consumed by AFB is approximately 27% of the total power. The filter bank designed for use in hearing aids can be classified as uniform filter banks and non-uniform filter banks different types offilter bank for reader’s reference. A 32-band discrete Fourier transform (DFT)filter bank was designed, and an 8-bandfilter bank with equal spaced finite-impulse response (FIR)filters was reported.

Non-uniformfilter banks can be further classified into octave band, critical-band, symmetric-band, and 1/3-octave-band filter banks. A 7-band octave filter bank was designed using the interpolated FIR (IFIR)filter technique. Lian and Wei proposed an 8-band octave filter bank with the IFIR and frequency-response masking (FRM) techniques to reduce the computational complexity. Chonget al. designed a critical-bandfilter bank to match well human perception. However, the irregular property of the critical bands makestheir implementation difficult.

In addition to, Wei and Lian proposed a 16-band symmetricfilter bank that guarantees high frequency resolution at both high and low frequency regions but rather low resolution near to .Kuoet al. recently proposed an efficient 18-band ANSI S1.11 1/3-octave filter bank . This 1/3-octave filter bank is suitable for both.

VI. ADVANTAGES

- To improve the output signal reconstruction effectively.
- To reduce the latency level and to increase the application system speed level.
- To optimize the overall power consumption level.

VII. PROPOSED DCT TRANSFORM

The proposed system is used to denoise the image using the simulink with XILINX blocks based design. This type of architecture is to combined the two domains. The proposed work is to modify the regular filtering process in input noisy speech signal and to analysis the internal high pass and low pass filtering samples.

The proposed speech signal filter architecture work is to convert the simulink design into hardware code & to optimize the filtering processing time.

A. *Design of DCT*

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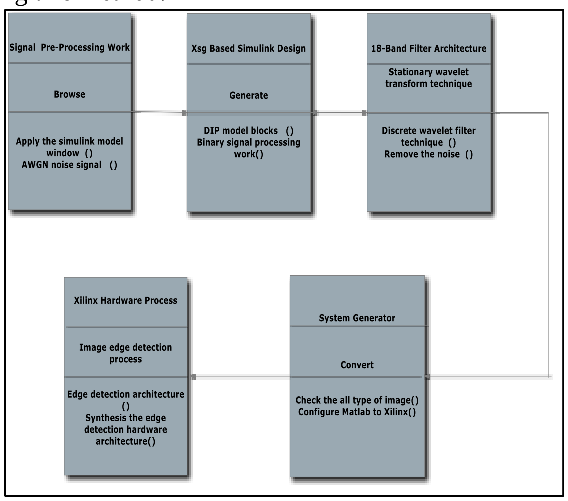

Fig. 1:Class Diagram

B. *18-Band Filter Architecture*

The simulink filter design is to follow the STATIONARY WAVELET transform technique and to minimize the filter circuit process.The discrete wavelet filter technique is consist the signal decomposition, thresholding technique and signal reconstruction block. The xilinx simulink design is used to remove the noise in given input speech signal and we use the 9-band low pass filter and high pass filter combination in signal decomposition side.

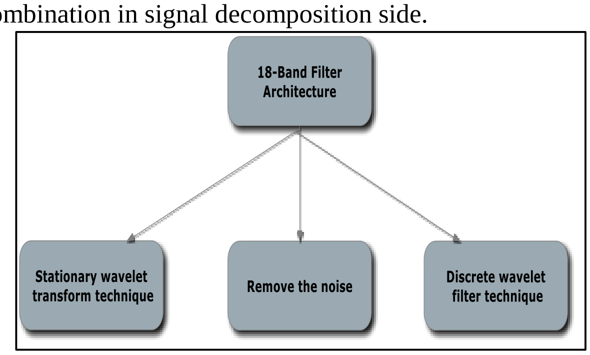

Fig. 2: 18-Band filter architecture

C. *System Generator*

The system generator process is used to convert the required Xilinx simulink model design into HDL programming language in Xilinx software.The system generator work is to check the all type of image processing Xilinx block and to create the hardware block package function.The system generator is simulation tool and to configure the Matlab to Xilinx process.

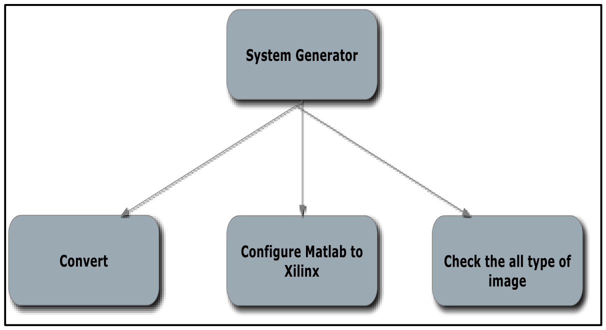

Fig. 3:System generator

D. Xilinx Hardware Process

To create the VHDL code for binary based image edge detection process.To calculate the complexity, power and time performance for edge detection architecture.Finally we simulate and synthesis the edge detection hardware architecture.

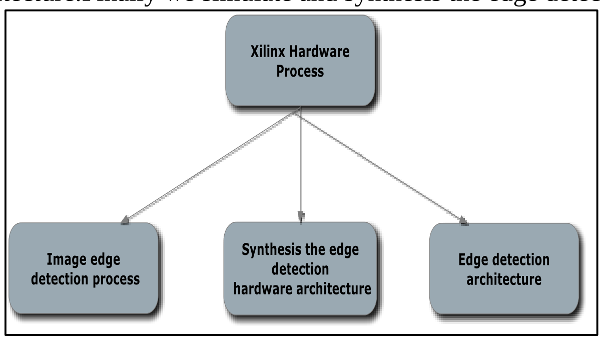

Fig. 4: Xilinx hardware process

VIII. SIMULATION RESULTS

Xilinx ISE (Integrated Software Environment) is a software tool produced by Xilinx for synthesis and analysis of HDL designs, enabling the developer to synthesize ("compile") their designs, perform timing analysis, examine RTL diagrams, simulate a design's reaction to different stimuli, and configure the target device with the programmer.The simulation results of the output waveforms in Xilinx are:

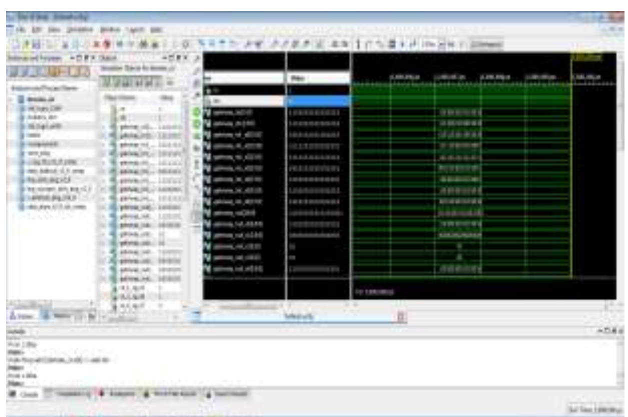

Fig. 5: Output Data

Table - 1

PARAMETER	*EXISTING SYSTEM*	*PROPOSED SYSTEM*
slice	*158*	*93*
lut	*133*	*39*
flip flop	*230*	*102*
speed(mhz)	*120.769*	*1171.78*
latency(ns)	*5.593*	*0.972*
delay(ns)	*8.280*	*0.853*
power	*1.7*	*0.102*

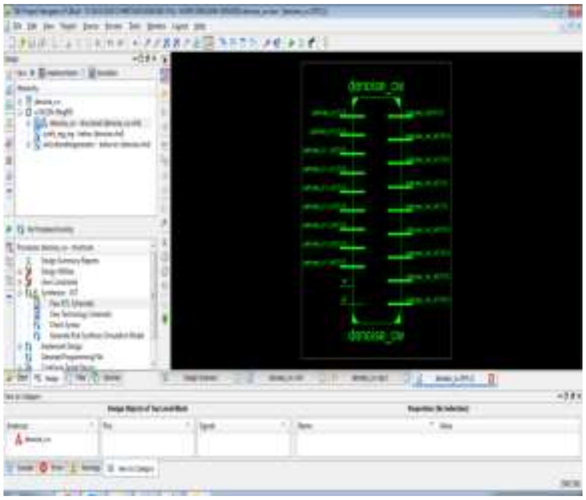

Fig. 6: RTL View of Proposed System

Fig. 7: Technology View of Proposed System

IX. PERFORMANCE ANALYSIS

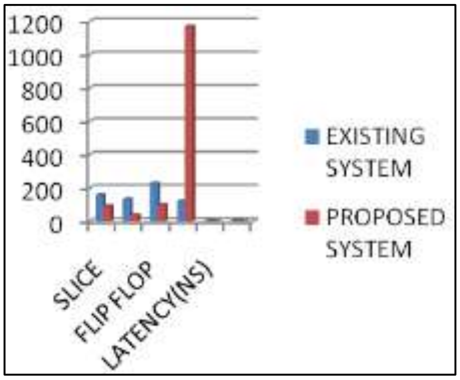

Fig. 8: Performance Graph

X. CONCLUSION

This proposed work provides the major advantage of the proposed design had fewer multiplications, fewer coefficients, less group delay, and less matching error than recent approaches. The implementation results indicated that the proposed DCT modulation accelerator can be operated at 95 MHz which is easily to achieve the real-time requirement. Overall, the proposed method would be more suitable for future system integrations and hearing aid applications.

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