Implementation of Quadrature Mirror Filter for Subband Adaptive Equalizer Application on FPGA

J. Sandhyarani, MRS. M. A. Asima begum, CH. Madhan mohan

Abstract— **In this decade the wireless communication technologies are expected to grow multifold and spreading its usage in all communication segments. The latest processing techniques are enabling the communication system to work with longer distances, with less energy per bit. The channel equalization is an important step in all most all wireless communication receiver designs.**

The Quadrature Mirror Filter (QMF) basically is a parallel combination of a High Pass Filter (HPF) and Low Pass Filter (LPF), which performs the action of frequency subdivision by splitting the signal spectrum into two spectra. By this QMF finds wide applications in many signal processing tasks such as trans-multiplexing, equalizing wireless communication channels; sub-band coding of speech and image signals, sub-band acoustic echo cancellation etc.

In this paper, hardware implementation of QMF is carried out on FPGA platform. VHDL will be used for RTL coding of the modules. The Xilinx IP Core generator will be used for instantiating the standard Xilinx parts. Xilinx ISE will be used to carryout the synthesis and bit file generation. The obtained Synthesis Report for implemented QMF will be used to analyze the occupied area and power dissipation. The study and implementation will be aimed to realize the equalizer for wireless communication system.

Index Terms— **Quadrature mirror filter (QMF), Equalizer, VHDL, FPGA.**

I. INTRODUCTION

Over the past two decades, the design of filter banks has received considerable attention in numerous fields such as speech coding, scrambling, image processing etc. [1]. Among the various filter banks, two-channel QMF bank was the first type of filter bank used in signal processing applications for separating signals into subbands and reconstructing them from individual subbands. Subsequently, a substantial progress has been made in other fields like antenna systems, Analog to Digital (A/D) converter, and design of wavelet based, due to advances in QMF bank. There are three types of distortions in QMF banks: aliasing distortion, phase distortion and amplitude distortion[2]. The aliasing distortion is eliminated with the use of suitable design of the synthesis filters and phase distortion is removed with the help of a linear phase FIR filter. Amplitude distortion can be minimized using computer aided techniques or can be equalized by cascading with a filter. Johnston has introduced the concept of a two band linear phase QMF banks. In linear phase QMF banks, aliasing and phase distortions are removed by choosing the analysis /synthesis filters and are assumed to have linear phase with even length. A nonlinear and iterative approach is used to minimize the amplitude distortion. The objective function is formulated using weighted sum of the ripples in the system response [3].

QMF banks are essential building blocks of a subband/transform coder. Generally, one wants the filter bank to have the perfect reconstruction (PR) property, which means that, in absence of coding, a delayed copy of the original signal is reconstructed exactly in the critically-sampled analysis-synthesis system. Although there are useful filter banks that are not exactly PR, if a PR filter bank with a fast implementation can be found, it is usually preferable. General conditions for PR have been derived by Vaidyanathan[4]. The below figure 1 shows the general structure of two-channel QMF bank.

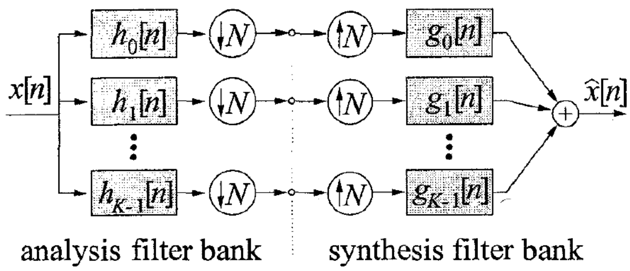

Fig. 1 Two-channel QMF bank

Although the DFT filter banks are widely used, there is a problem with aliasing in the decimated channels. At first glance, one might think that this is an insurmountable problem and must simply be accepted. Clearly, with FIR filters and maximal decimation, aliasing will occur. However, a simple example will show that it is possible to exactly cancel out aliasing under certain conditions[5].

II. EQUALIZATION

Equalization compensates for inter symbol interference (ISI) created by multipath within time dispersive channels. If the modulation bandwidth exceeds the coherence bandwidth of the radio channel, ISI occurs and modulation pulses are spread in time. An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics. Equalizers must be adaptive since the channel is generally unknown and time varying [6].

Inter symbol interference (ISI) caused by multipath in band limited (frequency selective) time dispersive channels distorts the transmitted signal, causing hit errors at the receiver. ISI has been recognized as the major obstacle to high speed data transmission over mobile radio channels. Equalization is a technique used to overcome inter symbol interference[7].

In a broad sense, the term equalization can be used to describe any signal processing operation that minimizes ISI. In radio channels, a variety of adaptive equalizers can be used to cancel interference while providing diversity, since the mobile fading channel is random and time varying, equalizers must track the time varying characteristics of the mobile channel, and thus are called adaptive equalizers[8].

III. IMPLEMENTATION

The Quadrature Mirror Filter (QMF) basically is a parallel combination of a High Pass Filter (HPF) and Low Pass Filter (LPF), which performs the action of frequency sub division by splitting the signal spectrum into two spectra. Thus QMF finds wide applications in many signal processing tasks such as trans-multiplexing, equalizing wireless communication channels; sub-band coding of speech and image signals, sub-band acoustic echo cancellation etc. The QMF consists of analysis filter bank and synthesis filter bank.

A. Analysis Filter bank

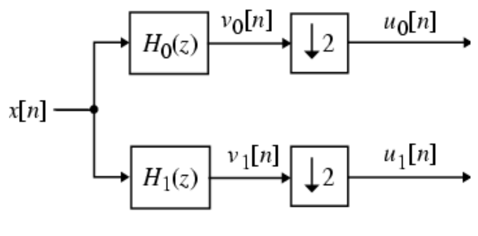

Fig. 2 Analysis filter bank

In Analysis filter bank, wide band of frequency will be divided into many sub-bands and those sub-bands are then subjected to equalization. The divided sub-bands are then combined at the synthesis filter bank.

The analysis filter bank is composed by a collection of M filters ("analysis filters", "decimation filters") with a common input signal. From the above figure 2, the first analysis filter $H_0(z)$ is usually of low-pass type and the other filters are of band pass type. The ideal of implementation but non-practical due to the filters properties is a bandwidth divided by perfect filter frequency responses. However, the practical implementation of the flters leads to a non-perfect eliminated band in their frequency responses which can produce several kinds of overlapping among them. This overlapping can be avoided with a correct design of the analysis filter bank.

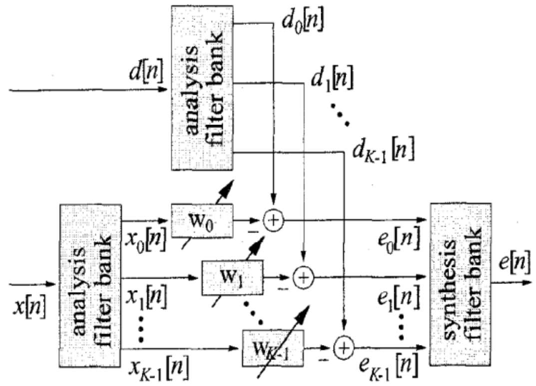

Fig. 3 Subband Filter

In the above figure 3, X[n] is the transfer function of signal received and is passed through the equalizer section. In equalizer section the whole frequency band is divided into number of sub bands according to the requirement d[n] is the transfer function of the training data. Each sub band is then compared with the training data at respected time slots and error signal e[n] will be generated and is added with the step size 'U' the combined signal is fed back to the accumulator and multiplier section (i.e. passing through LMS algorithm) again the same process repeated until the error signal is minimized. The Low frequency and high frequency subbands generated are given to Channel filter.

B. Channel Equalization

In digital communication, an equalizer is a device that attempts to recover a signal transmitted through an ISI channel. In communication system, it may be a simple linear filter or a complex algorithm. There are several equalizer types which are Linear Equalizer, Decision Feedback Equalizer, Blind Equalizer, and Adaptive Equalizer. In this paper we are implementing this by using Adaptive Equalizer.

The adaptive equalizer adapts to the source sample-by-sample, typically trained initially with a pseudorandom sequence.

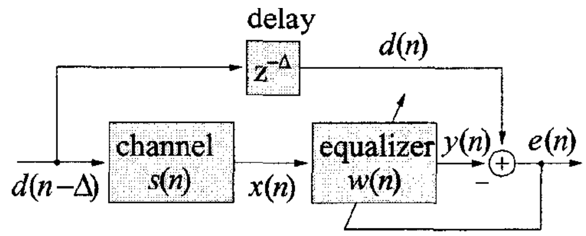

Fig. 4 Adaptive equalizer.

In this block diagram,

S(n) is the impulse response of the channel.

d(n-Δ) is the output of the transmitter .

x(n) is the output of the channel.

In the above figure 4, the signal which undergoes the channel whose impulse response is S(n) and the resulting signal is x(n). The y(n) and d(n) should only differ in delay because of noise in the channel. So, to introduce a special delay and compare them with two then we get an error e(n). Our goal is to minimize an error e(n) i.e. tending towards zero by tuning the equalizer this equalizer is called as Adaptive Equalizer.

C. Synthesis filter bank

After the signal is decomposed by the analysis bank, each signal is reconstructed using a synthesis bank structure. This structure is composed of filters and up-sampling that reconstructs each signal to its original sampling rate. Then the different components of the signal are quantized with the proper number of bits. In the end, the signal then needs to be reconstructed from the quantized individual components. First, the individual components are up-sampled (to undo the frequency-shift caused by down sampling) after which they pass through a bank of synthesis filters. The sum of the outputs of the synthesis filters is the output signal. The below figure 5 shows the synthesis filter bank.

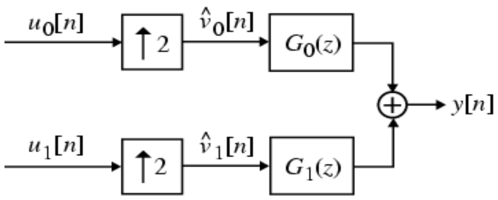

Fig. 5 Synthesis Filter bank

The immediate use for this method is the monitoring of time-varying individual power systems harmonics. Future use may include control and protection applications, as well as inter-harmonic measurements.

The synthesis filter bank gets output of the two DLMS blocks. These inputs are given to the synthesis filter bank in order to get the original input signal. This is reconstructed in the synthesis filter bank. The output of the block is named as "y_out".

IV. SIMULATION AND CHIPSCOPE RESULTS

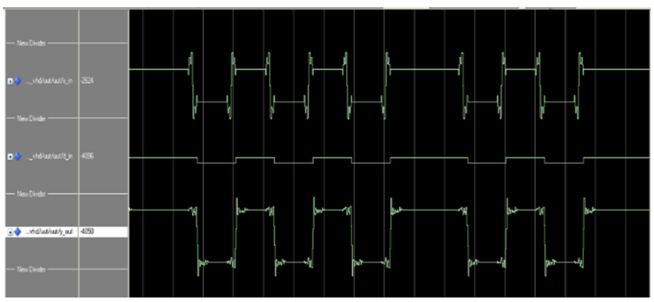

Fig. 6 Simulation results

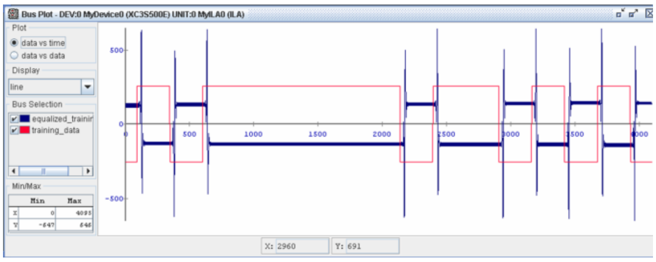

Fig. 7 Chipscope results

V. CONCLUSION

In this paper, we implemented the QMF for subband adaptive equalizer application of wireless communication channel in which the transmitted signal as been recovered using subband adaptive equalizer. The Modelsim Xilinx Edition (MXE) is used for simulation and functional verification. Xilinx ISE is used for synthesis and bit file generation. The Xilinx Chipscope is used to test the results on Spartan 3E 500K FPGA board.

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