Real Time Adaptive Parametric Equalization of Ultrasonic Transducers

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***Abstract*—Parametric equalization is often used to achieve a desired response from an audio transmitter, but is rarely applied to ultrasonic transducer systems. The ability of a broadband ultrasonic transmission and reception system to adapt its frequency and time domain response to changing acoustic conditions would be a distinct advantage in certain applications. Ultrasonic remote monitoring systems would benefit significantly from this ability, as signal levels could be minimized and consequentially the transmitter power consumption decreased. This work presents a real-time adaptive ultrasonic parametric equalizer using optimization driven Matlab code to control the coefficients of a switched capacitor filter network implemented in a Cypress PSOC (Programmable System On a Chip).**

In this work, adaptive parametric magnitude equalization of a through-transmission ultrasonic system using CUTs (Capacitive Ultrasonic Transducers) has been achieved in real time by tracking a desired SNR (signal to noise ratio) across the operational frequency spectrum. A Matlab general radial basis function (GRBF) artificial neural network (ANN) was developed to control the equalization filter coefficients based on the received frequency response data. The adaptive parametric equaliser adjusts the magnitude of the driving signal to maintain the desired SNR as closely as possible. The neural network was trained using PSO (Particle Swarm Optimization) back-propagation, based on a state space model of the system developed from frequency response data. The developed equalization circuitry, which is switched capacitor based and was fully implemented on the PSOC, is also described.

I. Introduction

In recent years, due to emerging requirements primarily in the fields of medical and industrial ultrasonic imaging capacitive ultrasonic transducers (CUTs) and capacitive micromachined ultrasonic transducers (CMUTs) have been receiving some attention as a potential alternative to piezoceramic devices in some applications. Much of this renewed attention is due to the increased potential of micromachined devices to support electric fields that allow CMUTs to compete with piezoelectric devices. In addition to being easily fabricated into large arrays and integrated into electronics [1], CUTs/CMUTs have a significantly greater bandwidth than piezoceramic devices. This increase in bandwidth not only allows for a significant increase in potential imaging resolution in medical ultrasonics but also facilitates optimal tissue harmonic imaging [2] which holds significant potential for future systems [3].

In all ultrasonic systems frequency dependent attenuation effectively means that as the acoustic separation distance is increased, the higher frequency elements of a signal become increasingly degraded. As most practical ultrasonic systems have changing channel parameters to a lesser or greater degree, this results in suboptimal frequency levels being used to ensure that SNR levels are adequate for worst case parameters. Static equalization filtering for audio channel compensation is well established [4] and has been explored previously for ultrasonics [5]. However, active channel compensation for ultrasonic systems would be a more useful technology. The use of standard active components using digitally programmable capacitors and resistors is an unattractive proposal as it does not hold any potential for integration onto an IC. The latency and required circuit architecture involved in digital channel equalization may become an issue in compact ultrasonic systems where space and cost is premium and the level of granularity offered by such a method is not required in most applications. As an alternative, switched capacitor filters are often used in the audio frequency range to circumvent the aforementioned difficulties, although they have been applied to the RF (radio frequency) spectrum [6] and for video application [7]. Additionally, VLSI implementations of these filters have been detailed [8] highlighting their ease of implementation and their application as equalizers is well understood [9].

Typically in adaptive channel equalization, FIR (Finite Impulse Response) is used as the fundamental architecture of the filter making algorithms such as LMS (Least Mean Squares) and RLS (Recursive Least Squares) easily implementable. However, FIR architectures tend to require significantly more computational power than a similar IIR (Infinite Impulse Response) counterpart. One of the key difficulties encountered when developing a high order IIR filtering circuits from arbitrary magnitude frequency domain data, as is the case in adaptive channel equalization, lies in the parameter determination, which is in itself an area of research [10]. This will be dealt with in more detail in Section II of this paper.

In Section II, an overview of the required theoretical framework is presented both in terms of switching capacitor circuits and neural networks. A detailed explanation of the experimental operation of the developed system is detailed in section III. Section IV then presents the results obtained from the system, followed by concluding comments in Section V.

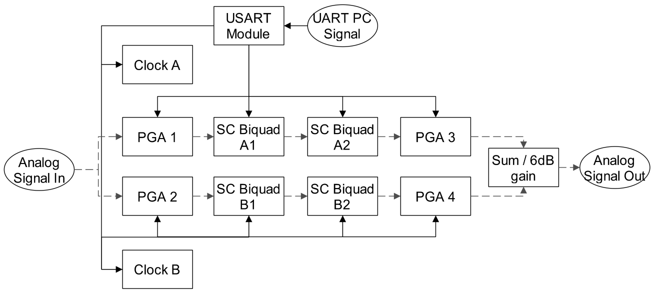

Fig. 1. Equaliser Design Block Level Layout

II. Theoretical Framework

A. Equaliser Design

The reconfigurable hardware of a Cypress PSOC contains architecture for SC bandpass biquadratic filters which are well suited to filtering sub 150kHz. However, SC filters have been shown to operate above this frequency range [6]. The frequency domain transfer function H, transferred from the time domain using the bilinear transformation, for a SC capacitor biquad of the type implemented in a PSOC is:

$$H = \frac{-\frac{C_1 C_B}{C_2 C_3} \frac{s(1+\frac{s}{sf_s})f_s}{\frac{C_A C_B}{C_2 C_3} - \frac{1}{4} - \frac{C_4}{2C_2}}}{s^2 + \frac{C_4}{C_2} \frac{sf_s}{\frac{C_A C_B}{C_2 C_3} - \frac{1}{4} - \frac{C_4}{2C_2}} + \frac{f_s^2}{\frac{C_A C_B}{C_2 C_3} - \frac{1}{4} - \frac{C_4}{2C_2}}} \tag{1}$$

where $s = j\omega$ and $j = \sqrt{-1}$, C_1, C_2, C_3 and C_4, are 5-bit dial-in filter capacitors, C_A and C_B are 1-bit dial-in filter capacitors and f_s is the sampling frequency. Consequentially gain G, center frequency f_c and q factor Q for this filter architecture may be defined as:

$$G = -\frac{C_1 C_B}{C_4 C_3} \tag{2}$$

$$f_c = \frac{1}{2\pi} \frac{f_s}{(\frac{C_A C_B}{C_2 C_3} - \frac{1}{4} - \frac{C_4}{2C_2})^{\frac{1}{2}}} \tag{3}$$

$$Q = \frac{C_2}{C_4} (\frac{C_A C_B}{C_2 C_3} - \frac{1}{4} - \frac{C_4}{2C_2})^{\frac{1}{2}} \tag{4}$$

The block level diagram of the system implemented using a Cypress PSOC can be seen in Fig 1. Firmware was written to listen indefinitely to the USART line for dial-in parameters and provide high level handshaking for the caller. When the parametric equalizer receives the dial-in values through the COM interface, it initially gathers and then sorts them. The parameters are then dialed into a pair of counters (used as the element clocks) and the required biquadratic blocks. The counters provide an 8-bit clock resolution and the capacitors have a 5-bit dial-in thus giving a combination space of 3.9x10^9 for each element, allowing approximation of a large number of desired coefficients.

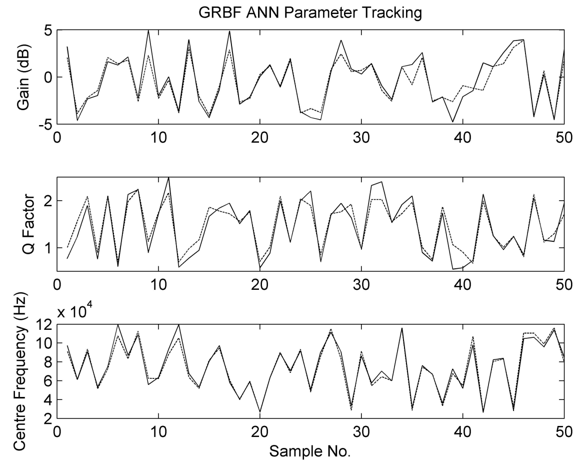

Fig. 2. GRBF ANN Parameter Tracking

B. Parameter Determination

The use of multiple equalizer elements to achieve a desired output is extremely difficult due to the large number of often conflicting parameters that must be selected. For example, optimizing parameters for an improved magnitude response may then result in degradation of the time domain response. Many parameters are indirectly coupled to each other, and the selection of a suitable set of parameters from desired frequency magnitude domain data is an ill-posed inverse problem that still attracts research interest [11]. In this work, in order to increase the speed of approximation of an arbitrary magnitude using a particular filter type, neural networks were used as an initial seed to fast minimization function. GRBF ANNs were found to be the most appropriate initial approximator for this form of problem as they can approximate any continuous function with arbitrary precision with enough hidden neurons. The output of a GRBF ANN $\varphi(x)$ may be given by:

$$\varphi(x) = \sum_{i=0}^{N} \alpha_i \rho(\|x - c_i\|) \tag{5}$$

Where α_i is the weight of the linear output neuron, N is the number of neurons and c_i is the center vector for the neuron. Fig 2 shows the parameter tracking on unseen data (gain, Q-factor and center frequency) for a four-pole Chebychev equalizer by a GRBF ANN utilizing PSO as its backpropogation method in the training set. The solid line shows the desired response, and the dashed line shows the corresponding output from the equalizer, showing that a very close match is achieved. The magnitude weighted spectrum error measure is then minimised through a quasi-newton function, which builds up curvature information at each iteration to formulate a quadratic model, which may be given by:

$$\min_x \frac{1}{2} x^T H x + c^T x + b \tag{6}$$

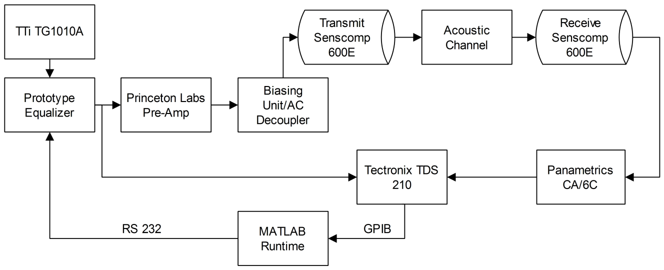

Fig. 3. Experimental Layout

where b and c are constant vectors, x is the minimization vector and H is the Hessian matrix. The optimum solution point x_{min} as such can then be expressed as:

$$x_{min} = -H^{-1}c \tag{7}$$

III. Experimental Operation

The parametric equalizer, implemented on a Cypress Semiconductors PSOC, was used to manipulate the response of a pair of SensComp 600 capacitive ultrasonic transducers with a nominal center frequency of 50 kHz operating in air. As previously detailed, an equalizer with two elements in parallel was implemented but more than two parallel elements are possible. A schematic of the experimental setup is shown in Fig 3. A TTi TG1010 arbitrary waveform generator (AWG) is used to produce the desired signal which then passes through the parametric equalizer and is amplified and coupled to a d.c. biasing circuit attached to the CUT transmitter. After propagating through an air gap, the ultrasound is detected by another Senscomp 600 transducer. The received signals are decoupled from the receiver bias, amplified through a Panametric CA/6C charge amplifier, digitized on a Tektronix TDS210 and transferred to a PC via a GPIB interface for analysis and parameter selection. The PC hosts the neural network and control algorithm used for real-time parameter selection and returns these parameters via a RS232 link to the Equalizer. The overall parametric equalizer operation is described by Fig 4. Upon initialization, the equalization controller code dials in capacitor values to the PSOC, giving low Q filters centered around the principal frequency of the ultrasonic transducers used (i.e. 50 kHz for the Senscomp 600 devices). A received ultrasonic wave is obtained from the oscilloscope to provide the adaptation algorithm with an initial signal to this dial-in value. As the type of filter to implement in the individual blocks is predefined, the phase characteristics can now be linked directly to the magnitude response and thus can be ignored when calculating the desired response, greatly simplifying the problem. The adaptation algorithm is then executed as described by the flow chart shown in Fig 5.

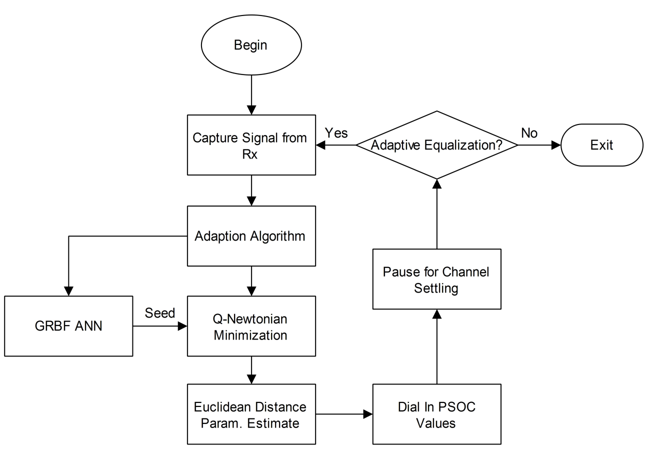

Fig. 4. Top Level Execution Loop Block Diagram

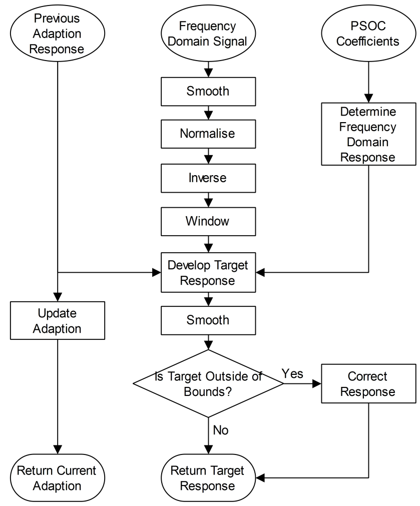

Fig. 5. Adaption Algorithm Flowchart

The desired response is then fed to a GRBF ANN as described previously. The GRBF ANN provides a seed value (i.e. initial estimate) to a quasi-newton minimization function that minimizes the generic parameters of the filter elements based on the cost function which is a weighted combination of center frequency, Q factor and gain. Having obtained these values for each element, the problem is then reduced to a solution space of $32 * 4 * N$ where N is the frequency divider, which was defined from an acceptable over sampling ratio (OSR). A direct search method is used on this reduced problem where the Euclidean distance measure is calculated for each point in the solution space. This calculation also ensures that invalid points (i.e. ones that return capacitor values outside the dial-in range) are not obtained and a weighting is given to the importance of each of the three parameters. From this distance measure, a matrix of the dial-in values is obtained for the closest match possible for a particular filter type. These parameters are then dialed-in through a COM(RS232) port and the cycle begins again with the carry-through parameter being the adaptation frequency domain response which is a cumulative measure of the deviation of the current response from the desired.

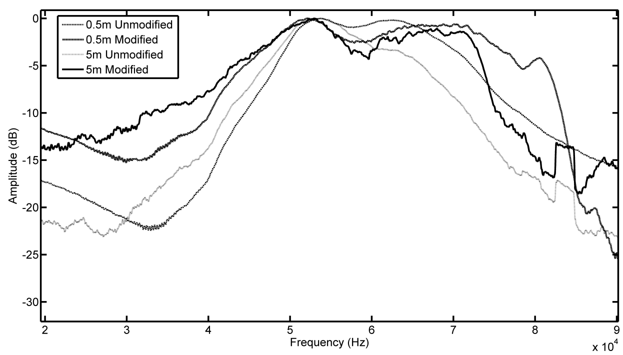

Fig. 6. Swept Sine Response Characteristic

IV. Results

Fig 6 shows the results for the goal of maximizing system 3dB bandwidth with increasing transducer separation distance. For clarity the extremes of the data set are shown, 0.5m seperation and 5m separation. Despite the progressive attenuation of higher frequencies with distance, the equalizer compensation can be seen to provide a similar system frequency response at these differing separations. A minimum increase in received signal 3dB bandwidth of almost 30% was observed. In the unmodified system, the 3dB bandwidth was seen to fall from 20.39kHz to 11.21kHz, while for the equalized system, the corresponding bandwidths were 28.22kHz and 26.13kHz. As a consequence, the bandwidth fluctuation across the distance set is reduced from 80% to 8%.

Experimental observation of the effect of channel equalization on an impulse response, transmitted across a 1m air gap, showed an increase in received pulse 6dB bandwidth by almost 160% as shown in Fig 7. The initial response without the equalizer is shown by the solid line and the received pulse 6dB bandwidth is approximately 25kHz. The received pulse with the equalizer is shown by the dashed line, where the pulse shape has been modified using a Bessel based equaliser with little elongation of the pulse length. The 6dB bandwidth of the pulse has been increased to approximately 65kHz.

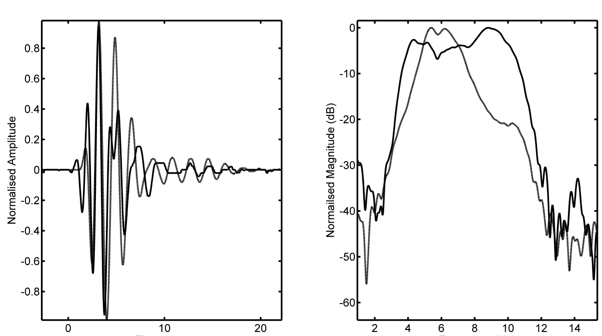

Fig. 7. Impulse Transient and Frequency Response Characteristic

V. Conclusion

The details of the implemented parametric equalization system have been outlined. It has been shown that for CUT through-transmission systems, adaptive switched capacitor parametric equalization may be used to compensate for changing frequency dependent attenuation in real time to maintain a desired SNR. Additionally, the initial results obtained for the modification of the transmit side of a pulsed ultrasonic system are promising. The principles of this work are valid for any transducer arrangement, frequency range, or equalization method. Perhaps the most promising potential application is the control of individual array elements in a phased array system, which is expected to allow for more robust operation. This work has begun to explore the application of adaptive filtering for air coupled ultrasonic systems of which there is significant potential.

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