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A Reliable Hardware Implementation Using Turbo Filters for Cochlear Implants

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Abstract

Cochlear implant (CI) is one of the best ways to restore hearing in the profoundly deaf candidates. CI typically comprises of crucial elements such as the microphone and the implant device. The implant device further comprises of electrodes which perform the equivalent task of hair cells. Researchers are constantly in a need of obtaining better algorithms to have enhanced speech for the receivers which could give a better quality of hearing. Also there is a need to validate the simulation results in hardware. Many signal processing algorithms are used. One useful technique is the Turbo method which makes use of time and frequency domain filters for its operation. The simulation studies for this algorithm are carried out and the performance for different noise types such as AWGN, Impulse and babble is analyzed. Once the results are obtained these results are validated by using a suitable hardware. The motivation for this work is to check the possibility of having reliable real time implementation in hardware. This work basically performs the validation of the results of the turbo code. It also gives the details of the experimental procedure for a hardware turbo implementation in real time making use of special features such as Software in loop (SIL).

Keywords: Cochlear Implants, Turbo, AWGN, Impulse, Babble, Software-in-loop (SIL)

1. Introduction

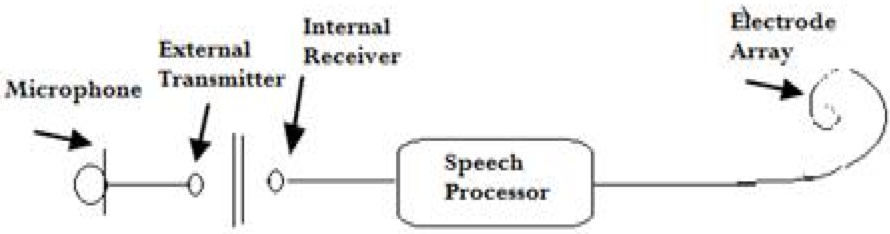

Fig 1 : Implantable system of a transcutaneous cochlear implant.[1]

Cochlear Implants (CI) are helpful to obtain hearing in profoundly deaf patients. Hearing is obtained by stimulating the auditory nerve of the patient. Stimulations are given by an array of electrodes. The success of the CI is due to lot of interdisciplinary expertise and further improvements are possible by considering the electrode design, signal processing techniques, materials used for electrode etc. Figure 1 represents an implantable system of a transcutaneous CI. It comprises of microphone, transmitter, receiver, speech processor and an array of electrodes. Future CI would be a fully implantable one with receiver and electrodes.[1]

Cochlear implants are very useful in cases of profound hearing losses. The success of CI could be mainly attributed due to the combined effort of many research areas such as bioengineering, physiology, otolaryngology, speech science and signal processing.[2]

Cochlear implant prosthesis is helpful to help hearing in profoundly deaf patients. Spatial electrode patterns help to obtain the hearing basically by making use of the auditory nerve.

Parameters such as number of electrodes, channels and the rate of stimulation are very important to get better recognition of sentence and words. Future implants would be required to consume very less power since they are fully worn inside the patient's body. [3]

Majority of the patients wearing the implants have high levels of speech recognition for hearing alone. Many of them can carry out telephonic conversations without major difficulty. Thus cochlear implant happens to be major achievements of modern age.[4]

A complete CI communication is carried by processing speech through different stages such as microphone , external speech processor, a transmission link with internal circuitry and an electrode or array of electrodes. Some of the important speech processing strategies could be compressed analog speech processor, continuous interleaved sampling processor, feature-based speech processors, Spectral Maxima sound processor etc. Research has proved that CIS and SMSP techniques are the best for multichannel/electrode CI. Some of the major issues of CI are that of size and the large amount of power used by the devices. To achieve compactness and low-power consumption features a device which makes use of high-density and low-power methods are required. Technologies such as MEMS may be useful in this regard.[5]

Real time implementation of CI could be done in many ways. One implementation is using Personal Digital Assistants (PDAs). The advantage of using a PDA was the availability of portable and cost effective computation platforms. Using dynamic link libraries, steps are carried out to obtain a real time throughput.[5]

Around 10% of the population in developed countries has serious hearing problems. There are two groups who suffer from hearing impairment. One group comprises of adult who are 65 years and above. Other group is of the age of 25 to 45 years, who are in the high risk of having an hearing impairment due to the their high risk working conditions. These may comprise roughly 42 % of probable candidates for hearing impairment.[5]

Implementation of speech processing based on auditory system and using continuous wavelet filtering module is done using TMS320C6416 DSP board. Two types of strategies were adopted viz making use of C language and also linear assembly based implementations.[6]

Software flexibility is a very useful attribute wherein we could deploy modified or new algorithms. Many real time implementations could be carried out successfully if development platforms are capable of possessing portability, flexibility and interactivity features.[7]

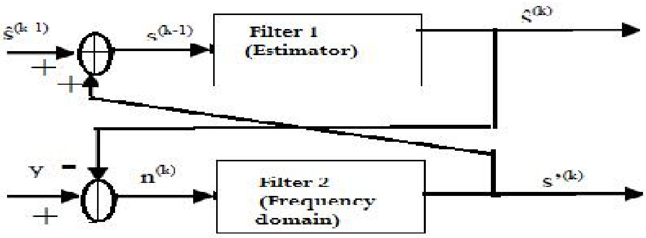

Fig 2. Turbo architecture [9]

Fig 2 shows the turbo architecture. Filter 1 is a time domain filter such as a Kalman filter and filter 2 is a frequency domain filter namely Dual Resonance Non-Linear filter. The turbo architecture makes use of the iterative process and it leads to an improved performance of each individual filter. The information from filter 2 is given to filter 1. Also the information of filter 1 is given to filter 2 as feedback. Initially the noisy speech signal is given as input to filter 1. This is basically an estimator which generates an estimate. This estimate is subtracted from the noisy signal which is further given as an input to the DRNL filter. The output obtained from DRNL is given to the previous kalman estimate. This serves as an input for the next iteration. The iterations are repeated to achieve maximum correlation coefficient compared to previous iterations. [8,9]

Implementation of a model in a code could be verified making use of Model-based design. Thus a transformation would result wherein a model is represented using a generated code.[10]

Modern microelectronic techniques are used to develop artificial cochlear implants. Future cochlear implants can mitigate the energy consumption by reducing the number of electrodes. [11]

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2. Methodology

The Turbo method is implemented initially to obtain the maximum performance in terms of maximum correlation between the input and the output. The next step is to validate this performance by making use of a set up which could further confirm the simulation results obtained earlier. For this we make use of a special feature referred to as software in loop. The details of the validation include the use of a Matlab Coder with prerequisites. The Inputs and Outputs are clearly defined by providing their types. All functions are cleansed to meet the Matlab Coder requirements. All dependencies on Matlab functions are resolved by creating custom functions that replace the in- built functions. The next step is to test the C Code. Testing the C code is accomplished through Software-in-Loop (SIL) testing. For this, a Matlab script is written that reads the input, calls the functions/modules that are under test and records the output. The Matlab SIL component reroutes the function calls to the C code and routes the results back into the test script. The real-time implementation is carried out on a BeagleBoard-xM ARM processor. Speech is recorded using electret microphones through an I/O library such as PortAudio. The filter algorithms process the incoming audio signal in real-time and record the results. The output can be written into audio files for obtaining parameter statistics, and is also routed to speakers for subjective analysis.

Implementation details : Using beagleboard-xM ARM processor running at 1 GHz. A single-channel audio input is fed using a condenser microphone. For capturing the microphone input and converting it suitably, PortAudio which is a cross-platform audio I/O library is used. Further the results are validated with the MATLAB. A special feature of this implementation was the SIL.This involved running the MATLAB script and routing the filter function calls to the c-implementations.

3. Results

Fig 2, gives the snapshot of the hardware implementation details of the Turbo method.

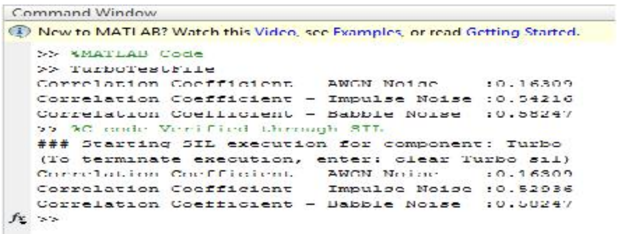

Fig 2: Validation for Turbo code using SIL.

Referring figure 2, it can be seen, the results of the C implementation match closely with the Matlab results, thus validating it. Correlations values are with respect to a single audio sample are displayed.

4. Analysis and future scope

Real-time Optimizations –Utilize patch for the OMAP linux kernel that is based on the rt patch (RT_PREEMPT). This patch allows true real-time behavior on hardware platform such as the BeagleBoard xM. Implementation is carried by considering recurring functions such as Butterworth filter as lookup tables with corresponding configuration values.

Arithmetic – ARM Cortex A8 provides NEON which is a 64/128-bit wide SIMD vector extension for ARM. It has been architected to be an efficient C compiler target as well as being used from assembly language. It has 32x 64-bit registers (with a dual view as 16x 128-bit registers) which can hold various commonly used datatypes.

The C code has been compiled with the NEON option enabled in the GCC compiler which automatically leverages NEON instruction set wherever possible to speed up the execution.

5. Conclusions

A reliable hardware implementation of DRNL filter is not available online. Using the SIL features enables to test the filter block in real-time. Enables us to develop and test algorithms in the future that take advantage of the stereo mics to further enhance the audio.

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