Computers and Electrical Engineering 74 (2019) 391–412

Contents lists available at ScienceDirect

Computers and Electrical Engineering

journal homepage: www.elsevier.com/locate/compeleceng

Low-complexity implementation of efficient reconfigurable structure for cost-effective hearing aids using fractional interpolation☆

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ARTICLE INFO

Article history:
Received 1 November 2017
Revised 9 February 2019
Accepted 11 February 2019

Keywords:
Fractional interpolation
Hearing aid
Filter bank
Reconfigurable
Multiplier-less
Implementation

ABSTRACT

Digital hearing aids can be made less expensive if they are reconfigurable and of low hardware complexity. Hence, this work proposes a hardware-efficient, reconfigurable filter bank structure, based on fractional interpolation. The proposed structure is reconfigurable since a single structure can be used for different patients with different types of hearing impairments. The proposed structure consists of a masking stage and a scheme generation stage with a prototype filter in each stage. The prototype filters are made multiplier-less by representing their coefficients in the CSD space. The filter characteristics are improved by deploying a MOABC optimization algorithm. The number of adders is reduced using the SIDC-CSE technique. The low-complexity structure can be used for all types of hearing impairments, with the matching error and delay within tolerable ranges. The proposed structure has been implemented on an FPGA to support the analytical results for low hardware complexity and hence low power.

1. Introduction

Digital hearing aids help to compensate for the hearing loss of a hearing-impaired person. In a hearing aid, sounds at all frequencies are amplified selectively and are transmitted to the ear. A block diagram of a digital hearing aid [1] is illustrated in Fig. 1. The analogue output of the microphone is digitized by an analogue-to-digital converter, shown as A/D, which is fed as the input to the filter bank. The filter bank block, consisting of a bank of filters and gain blocks, is a vital part of a digital hearing aid. The input signal is broken down into various sub-band components by the bank of filters. The gain block provides selective amplification to each of these sub-bands, according to the person's hearing ability. These sub-bands are additively combined, and then converted back into an analogue signal by a digital-to-analogue converter labeled as D/A in Fig. 1.

An audiogram is a graph which represents a person's hearing ability at different frequencies. A typical audiogram along with the necessary details of the hearing levels is shown in Fig. 2. A person's hearing level at different frequencies can be determined from the audiogram. According to the intensity values, hearing loss is categorized as normal (0 to 20 dB), mild (20 to 40 dB), moderate (40 to 70 dB), severe (70 to 90 dB) or profound (greater than 90 dB) [2]. Intensity measures are

☆ Reviews processed and recommended for publication to the Editor-in-Chief by Area Editor Dr. E. Cabal-Yepez.

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https://doi.org/10.1016/j.compeleceng.2019.02.008
0045-7906/© 2019 Published by Elsevier Ltd.