FPGA Based Implementation of Comb Filters for Use in Binaural Hearing Aids for Reducing Intraspeech Spectral Masking

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***Abstract*— Sensorineural hearing impairment is associated with decreased speech perception due to increased intraspeech spectral masking. For persons with moderate bilateral sensorineural loss, binaural dichotic presentation using a pair of complementary comb filters can reduce the effects of increased intraspeech spectral masking and thereby improve speech perception. It has been earlier shown that use of comb filters based on auditory critical bandwidths, with magnitude responses designed for perceptual balance of loudness and linear phase responses, resulted in a significant improvement in speech perception without adversely affecting localization of broadband sound sources. An FPGA-based implementation of these 513-coefficient filters with sampling frequency of 10 kHz is carried out for use in binaural hearing aids. Implementation using a 16-bit codec and 15-bit integer filter coefficients used 47%, 34%, and 53% of combinational functions, logic registers, and logic elements, respectively, available on "Altera Cyclone II EP2C70F896C6" FPGA. The resulting magnitude responses have a close match to the offline floating-point implementation.**

Keywords—binaural hearing aid; comb filters; FPGA based implementation; sensorineural hearing loss

I. INTRODUCTION

Hearing impairment can be classified, on the basis of location of the defect in the auditory system, as conductive, sensorineural, and central losses [1] – [3]. Conductive loss occurs due to an abnormality in the middle ear leading to poor transmission of the sound to the inner ear. Sensorineural loss is caused by pathology in the cochlea and/or due to degeneration of the auditory nerves. Central loss occurs due to inability of the brain in decoding the neural firings into meaningful linguistic information. Sensorineural loss is associated with elevated hearing thresholds, decreased dynamic range and loudness recruitment (abnormal loudness growth), and increased temporal and spectral masking, resulting in degraded speech perception.

Hearing aids generally provide frequency-selective amplification to compensate for the elevated hearing thresholds. Automatic volume control and multichannel dynamic range compression (with settable attack time, release time, and compression ratios) are used to partially address the problems associated with the reduced dynamic range and loudness recruitment. Increased temporal masking results in poor detection of acoustic events. Increased spectral masking caused by widening of auditory filters results in poor discrimination of spectral features. The increased masking makes speech perception very difficult in the presence of noise. It also results in poor speech perception due to increased intraspeech masking. Improvement of consonant-to-vowel ratio (CVR) has been investigated for reducing the effects of increased temporal masking [4], [5]. Techniques based on spectral contrast enhancement [6], [7] and multiband frequency compression [8], [9] have been used for reducing the effects of increased spectral masking. These techniques introduce processing related artifacts and have high computational requirements.

For persons having moderate bilateral sensorineural loss who can use binaural aids, the effects of increased intraspeech masking can be reduced by using comb filters with complementary magnitude responses [10] – [12]. The masking takes place primarily at the peripheral level, and integration of binaural information takes place at higher levels in the auditory system. In binaural presentation using complementary comb filters, the spectral components likely to mask or get masked by each other are presented to different ears, for reducing the adverse effect of increased intraspeech spectral masking thereby improving the speech perception.

Investigations reported in [12] showed that comb filters based on auditory critical bandwidths [13], with magnitude responses designed for perceptual balance of loudness and linear phase responses, resulted in a significant improvement in speech perception. The processing was evaluated by conducting modified rhyme test (MRT) [14], [15] on subjects with moderate bilateral sensorineural loss (average pure tone threshold: 40 – 77 dB HL, asymmetry < 12 dB) for a quantitative measure of speech intelligibility for words with both initial and final position variations of consonants. The test results showed 14 – 31% improvement in the consonant recognition scores and a decrease of 0.26 s in mean response time. It was further established that the presentation using these comb filters did not adversely affect localization of source direction for broadband environmental sound and speech. The paper presents implementation of these comb filters for use in binaural hearing aids.

Several investigations involving implementation of signal processing applications with highly parallel computations have reported FPGAs to be better suited than DSP chips [16] – [19]. The FPGA structure facilitates parallel and compound data processing operations every clock cycle, whereas the DSP-chip based processing involves sequential instruction fetch-and-execute cycles [16]. Because of power and space constraints, hearing aids are generally designed using ASIC (application specific integrated circuit), which involves significant nonrecurring cost related to chip fabrication. This cost can be avoided by using FPGA for fast prototyping [20] – [22] before taking the design to ASIC stage. It provides flexibility especially when targeting designs with low-leakage currents in deep submicron chips [23], [24]. RTL-to-GDS is a fully automated ASIC digital design flow and is supported by most of the VLSI tool vendors. It can be used for transforming synthesized VHDL or Verliog code into GDSII mask layout file [25] – [27].

As filtering can be realized using a parallel computation structure, an FPGA platform is used for implementing the comb filters for use in binaural hearing aids. After performance verification, the FPGA-based design can be converted to ASIC design.

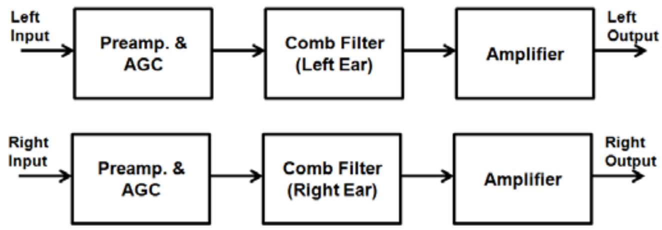

Fig. 1. A binaural hearing using comb filters

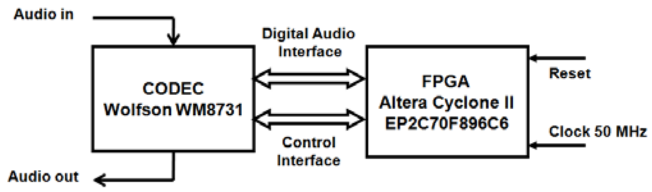

Fig. 2. FPGA board for comb filter implementation

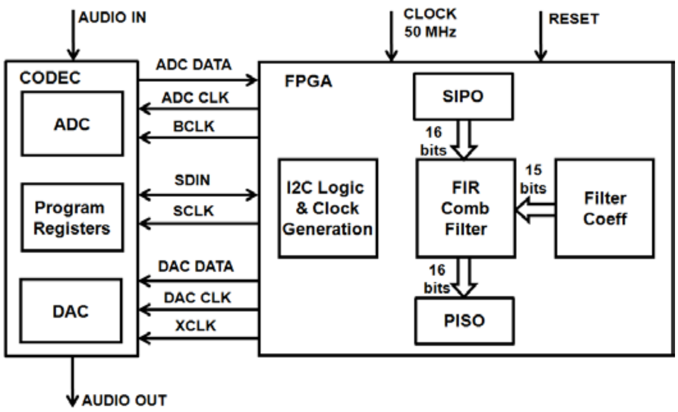

Fig. 3. Comb filter implementation

II. FPGA BASED IMPLEMENTATION

Figure 1 shows block diagram of a binaural hearing aid using comb filters for dichotic presentation to reduce the adverse effects of increased intraspeech spectral masking. The filters used in [12] were designed as linear-phase FIR filters with the magnitude responses meeting the requirements of narrow transition widths, low pass-band ripple, large stop-band attenuation, and very small deviation in the sum of the magnitude responses on a linear scale. The filters were designed using iterative application of frequency sampling method of filter design [28], [29]. For the sampling frequency of 10 kHz, the 513-coefficeint filters were realized using Matlab-based offline processing with floating-point operations. We have used the same set of filter coefficients, scaled to 15-bit sign magnitude integers for FPGA-based implementation. These FIR comb filters were implemented using “Altera DE2-70” board. Figure 2 shows the interconnection of FPGA “Altera Cyclone II EP2C70F896C6” and audio stereo codec “Wolfson WM8731” on the board used for implementing the filter.

Figure 3 shows details of implementation. The codec has input and output range of 1 V RMS. One channel of the stereo codec is programmed using I2C control interface for 16-bit analog-to-digital and digital-to-analog conversion at sampling frequency of 10 kHz. The filtering operation takes place using the filter coefficient values from the registers. The resulting 34-bit output is scaled to 16-bit and output through DAC.

The codec has four audio interface modes: left justified, right justified, DSP, and I^2S. All modes use 2's complement data with 16/20/24/32 bits. Implementation was carried out using left justified mode. In this mode, the MSB is available on the first rising edge of BCLK following the ADCLR or DACLRC transition. The codec operation is controlled by writing to its ten registers. For this purpose, a two-wire I2C (SDIN and SCLK) controller was implemented in FPGA as per the timing specification of the codec. Writing to each register involves transfer of 16 bits (7 address bits and 9 data bits, sent as two bytes). To write to a register of the codec, the controller establishes a high-to-low transition on SDIN while SCLK is high and then sends the device address of the codec. After an acknowledgement from the codec by pulling SDIN low for one clock cycle, the controller sends the first 8 bits. After receiving acknowledgement, the controller sends the remaining 8 bits. After acknowledgement from the codec, the controller indicates a stop condition by low-to-high transition on SDIN while SCLK is high. If start or stop condition is detected out of sequence during data transfer, the device goes to idle condition.

Coefficient scaling and filter realization were carried out with the help of Altera MegaWizard plug-in manager FIR compiler 9.1. Out of the several filter structures supported by the compiler, variable/fixed coefficient multi-cycle architecture was selected as it provides high throughput needed for real-time processing in hearing aids. Processing is carried out sample-by-sample with 513-tap filtering, involving 513 multipliers and registers along with other associated logic. Implementation of the comb filter used about 47%, 34%, and 53% of combinational functions, logic registers, and logic elements, respectively of the FPGA device.

III. RESULTS

The implementation was tested for realizing comb filters using scaled integer coefficients for left ear and right ear. The frequency responses of the two comb filters were measured using swept tone as the input. The filters operated in linear region for sinusoidal inputs up to 0.25 V RMS. For a comparison, magnitude responses of the two comb filters with offline processing using floating point operations as reported in [12] are shown in Figure 4. These filters have complementary magnitude responses with narrow transition widths (< 55 Hz), low pass-band ripple (< 1 dB), large stop-band attenuation (>30 dB), and cross-over gains of -5 to -6 dB. The deviation in the sum of the magnitude responses on a linear scale is very small (< 0.04). Figure 5 shows the responses of the FPGA-based implementation. The responses have transition widths of less than 75 Hz. The pass-band ripples are below 2 dB, stop band attenuations are greater than 18 dB, and cross-over gains are -4 to -8 dB. Deviation in the sum of magnitude responses on a linear scale is below 0.098. Thus the properties of the FPGA-based comb filters closely match with those of the filters in [12]. Figure 6 shows the spectrograms of an input speech signal and two outputs of the FPGA-based comb filters showing complementary splitting of the spectra. Binaural presentation through headphones of the test stimuli used for listening tests in [12] did not show any perceptual distortion for the processed sounds indicating nearly perfect perceptual fusion of the binaural sounds.

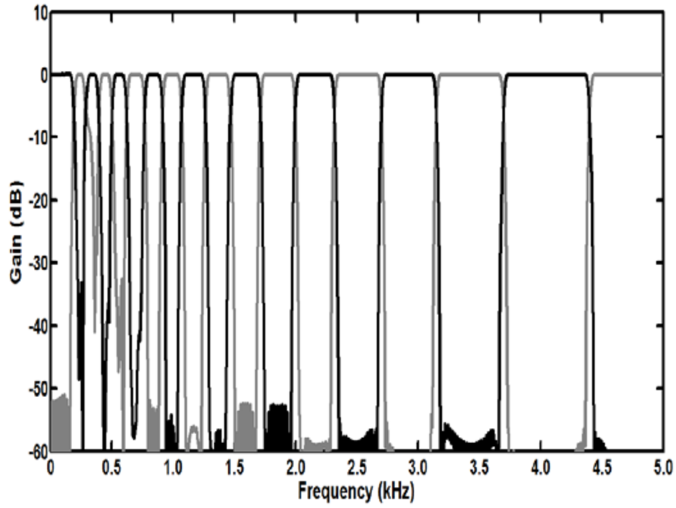

Fig. 4. Magnitude responses of comb filters for offline processing as used in [12]

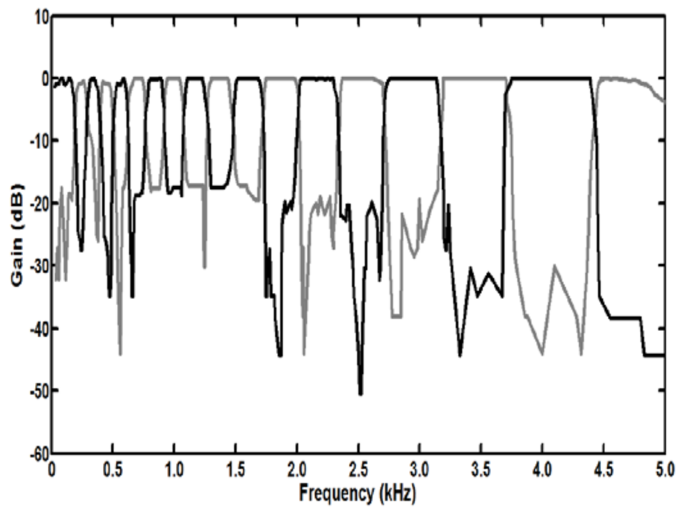

Fig. 5. Magnitude responses of comb filters implemented using FPGA

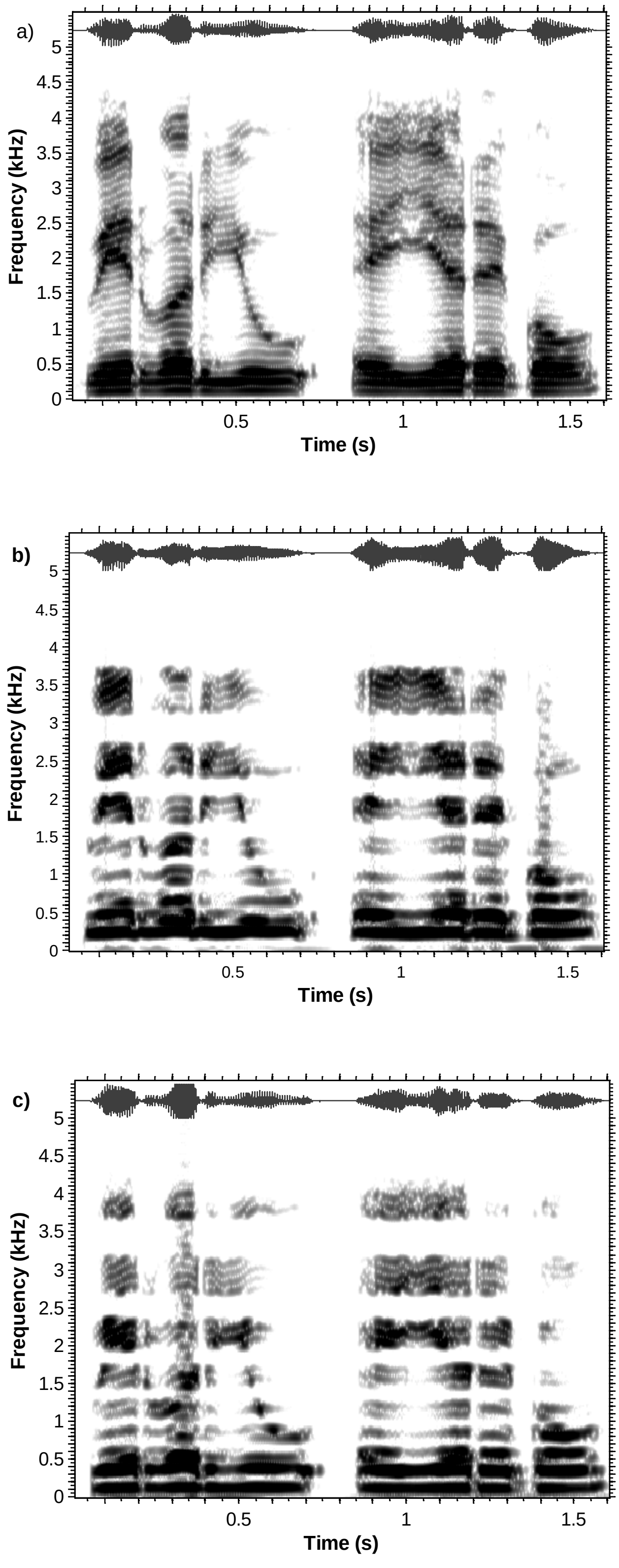

Fig. 6. Wide band spectrograms of the sentence "where were we a year ago" from a male speaker: a) unprocessed, b) output of left ear comb filter, c) output of right ear comb filter

IV. Conclusion

An earlier investigation [12] has shown that a pair of comb filters based on auditory critical bandwidths, with magnitude responses designed for perceptual balance of loudness and linear phase responses could be useful in improving speech perception by listeners with moderate bilateral sensorineural loss. For possible use in binaural hearing aids, these comb filters were implemented using an FPGA board and tested. Implementation using a 16-bit codec and 15-bit filter coefficients resulted in satisfactory frequency response and used 47%, 34%, and 53% of combinational functions, logic registers, and logic elements, respectively, available on FPGA “Altera Cyclone II EP2C70F896C6”. The implementation has to be integrated with automatic gain control and filtering with frequency-selective response to compensate for the elevation of hearing thresholds. It needs to be tested on listeners with sensorineural loss and using binaural hearing aids.

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