FPGA Implementation of Hearing Impaired Assistive Device for Hard to Hear Individuals

T.Aswathi
Electronics & Communication Engineering,
Shivani Engineering College, Trichy, Tamil Nadu, India.

Abstract—The Noise cancellation and suppression techniques have been developed and implemented in FPGA in this work. Hearing aids are primarily meant for improving hearing and speech comprehensions. Digital hearing aids score over their analog counterparts. This happens as digital hearing aids provide flexible gain besides facilitating feedback reduction and noise elimination. Recent advances in digital signal processors (DSP) and Microelectronics have led to the development of superior digital hearing aids. Many researchers have investigated several algorithms suitable for hearing aid application that demands low noise, feed-back cancellation, echo cancellation, etc., however the toughest challenge is the implementation. Furthermore, the additional constraints are power and area. The device must consume as minimum power as possible to support extended battery life and should be as small as possible for increased portability. In this work, we are using cross-channel suppression technique to remove the unwanted audio signals. The unwanted signals are suppressed using two-tone suppression scheme. In this project, the speech signal is absorbed by microphone. This signal is then converted to digital using ADC. The digitized signal is processed using FPGA. Here in FPGA the speech signal is enhanced and amplified to the desired level. The processed speech signal is then converted into analog format using DAC and is given to speaker.

Index Terms—Hearing Impaired, personal assistance device, FPGA

I. Introduction

A hearing aid is a small electronic device that one wears in or behind his/her ear. It makes sounds louder so that a person with hearing loss can listen, communicate, and participate better in daily activities. A hearing aid can help people hear more in both quiet and noisy situations. A hearing aid has three basic parts: a microphone, amplifier, and speaker. The hearing aid receives sound through a microphone, which converts the sound waves to electrical signals and sends those to an amplifier. The amplifier increases the power of the signals and then sends to the ear through a speaker [1-5]. Hearing aids are primarily useful in improving the hearing and speech comprehension of people who have hearing loss that results from damage to the small sensory cells in the inner ear, called hair cells. The damage can occur as a result of disease, aging, or injury from noise or certain medicines. A hearing aid magnifies sound vibrations entering the ear. Surviving hair cells detect the larger vibrations and convert them into neural signals that are passed along to the brain. The greater the damage to a person's hair cells, the more severe is the hearing loss, and the greater the hearing aid amplification needed to be larger. How-ever, there are practical limits to the amount of amplification a hearing aid can provide. In addition, if the inner ear is too damaged, even large vibrations will not be converted into neural signals. In this situation, a hearing aid would be ineffective [6-8].

Corresponding author: aswathiece94@gmail.com

The Noise cancellation and suppression techniques have been developed and implemented in FPGA in this dissertation. Hearing aids are primarily meant for improving hearing and speech comprehensions. Digital hearing aids score over their analog counterparts. This happens as digital hearing aids provide flexible gain besides facilitating feedback reduction and noise elimination. Recent advances in DSP and Microelectronics have led to the development of superior digital hearing aids [9-14]. Many researchers have investigated several algorithms suitable for hearing aid application that demands low noise, feedback cancellation, echo cancellation, etc., however the toughest challenge is the implementation. Furthermore, the additional constraints are power and area. The device must consume as minimum power as possible to support extended battery life and should be as small as possible for increased portability. In this work, we are using cross-channel suppression technique to remove the unwanted audio signals. The unwanted signals are suppressed using two-tone suppression scheme. In this project, the speech signal is absorbed by microphone. This signal is then converted to digital using ADC. The digitized signal is processed using FPGA. Here in FPGA the speech signal is enhanced and amplified to the desired level. The processed speech signal is then converted into analog format using DAC, it is given to speaker [15-20].

In this work we proposed as cross-channel suppression technique to remove the noisy audio signals. The unwanted signals are suppressed using two-tone suppression scheme. In this work, the speech signal is absorbed by microphone; this signal is then converted to digital format using ADC. The digitized signal is processed using FPGA. Here in FPGA the speech signal is enhanced and amplified to the desired level. The processed speech signal is then converted into analog format using DAC and will be given to speaker.

II. The proposed system

Varieties Microphones are referred to by their transducer principle, such as condenser, dynamic, etc., and by their directional characteristics. Sometimes other characteristics such as diaphragm size, intended use or orientation of the principal sound input to the principal axis (end- or side-address) of the microphone are used to describe the microphone. A microphone, colloquially mic is an acoustic-to-electric transducer or sensor that converts sound in air into an electrical signal[21-23]. Most microphones today use electromagnetic induction (dynamic microphones), capacitance change (condenser microphones) or piezoelectricity (piezoelectric microphones) to produce an electrical signal from air pressure variations. Microphones typically need to be connected to a preamplifier before the signal can be amplified with an audio power amplifier. Then the audio signal is given to the analog to digital converter after converting into digital the data will be given to the FPGA module, here the signal noise is suppressed using cross-Channel Suppression. The output from the FPGA is converted into analog signal using digital to analog converter the output is given to the speaker.

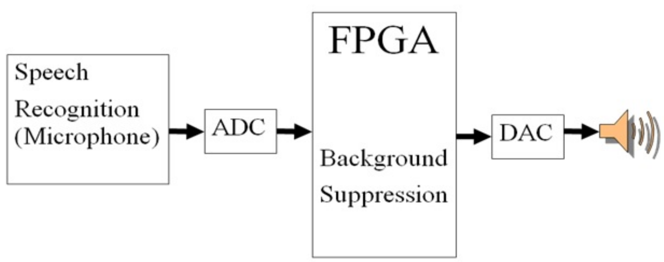

Fig. 1. Experimental setup for the proposed system.

A. Microphone

The sensitive transducer element of a microphone is called its element or capsule. Except in thermo phone based microphones, sound is first converted to mechanical motion by means of a diaphragm, the motion of which is then converted to an electrical signal. A complete microphone also includes a housing, some means of bringing the signal from the element to other equipment, and often an electronic circuit to adapt the output of the capsule to the equipment being driven. A wireless microphone contains a radio transmitter. The condenser microphone, invented at Bell Labs in 1916 by E. C. Wente is also called a capacitor microphone or electrostatic microphone—capacitors were historically called condensers. Here, the diaphragm acts as one plate of a capacitor, and the vibrations produce changes in the distance between the plates. There are two types, depending on the method of extracting the audio signal from the transducer: DC-biased microphones, and radio frequency (RF) or high frequency (HF) condenser microphones. With a DC-biased microphone, the plates are biased with a fixed charge (Q). The voltage maintained across the capacitor plates changes with the vibrations in the air, according to the capacitance equation ($C = Q/V$), where Q = charge in coulombs, C = capacitance in farads and V = potential difference in volts. The capacitance of the plates is inversely proportional to the distance between them for a parallel-plate capacitor. (See capacitance for details.) The assembly of fixed and movable plates is called an "element" or "capsule". A nearly constant charge is maintained on the capacitor. As the capacitance changes, the charge across the capacitor does change very slightly, but at audible frequencies it is sensibly constant. The capacitance of the capsule (around 5 to 100 pF) and the value of the bias resistor (100 MΩto tens of GΩ) form a filter that is high-pass for the audio signal, and low-pass for the bias voltage. Note that the time constant of an circuit equals the product of the resistance and capacitance. Within the time-frame of the capacitance change (as much as 50 ms at 20 Hz audio signal), the charge is practically constant and the voltage across the capacitor changes instantaneously to reflect the change in capacitance. The voltage across the capacitor varies above and below the bias voltage. The voltage difference between the bias and the capacitor is seen across the series resistor. The voltage across the resistor is amplified for performance or recording. In most cases, the electronics in the microphone itself contribute no voltage gain as the voltage differential is quite significant, up to several volts for high sound levels. Since this is a very high impedance circuit, current gain only is usually needed, with the voltage remaining constant.

B. FPGA

A field-programmable gate array (FPGA) is an integrated circuit designed to be configured by a customer or a designer after manufacturing – hence "field-programmable". The FPGA configuration is generally specified using a hardware description language (HDL), similar to that used for an application-specific integrated circuit (ASIC) (circuit diagrams were previously used to specify the configuration, as they were for ASICs, but this is increasingly rare). FPGAs contain programmable logic components called "logic blocks", and a hierarchy of reconfigurable interconnects that allow the blocks to be "wired together" – somewhat like many logic gates that can be inter-wired in different configurations. Logic blocks can be configured to perform complex combinational functions, or merely simple logic gates like AND and XOR. In most FPGAs, the logic blocks also include memory elements, which may be simple flip-flops or more complete blocks of memory [24]. Traditionally, FPGAs have been reserved for specific vertical applications where the volume of production is small. For these low-volume applications, the premium that companies pay in hardware costs per unit for a programmable chip is more affordable than the development resources spent on creating an ASIC for a low-volume application. Today, new cost and performance dynamics have broadened the range of viable applications.

C. FPGA Design and Programming

To define the behaviour of the FPGA, the user provides a hardware description language (HDL) or a schematic design. The HDL form is more suited to work with large structures because it's possible to just specify them numerically rather than having to draw every piece by hand. However, schematic entry can allow for easier visualisation of a design. Then, using an electronic design automation tool, a technology-mapped netlist is generated. The netlist can then be fitted to the actual FPGA architecture using a process called place, usually performed by the FPGA Company's proprietary place-and-route software. The user will validate the map, place and route results via analysis, simulation, and other verification methodologies. Once the design and validation process is complete, the binary file generated (also using the FPGA company's proprietary software) is used to (re)configure the FPGA. This file is transferred to the FPGA/CPLD via a serial interface (JTAG) or to an external memory device like an EEPROM. The most common HDLs are VHDL and Verilog, although in an attempt to reduce the complexity of designing in HDLs, which have been compared to the equivalent of assembly languages, there are moves to raise the abstraction level through the introduction of alternative languages. National Instruments' Lab VIEW graphical programming language (sometimes referred to as "G") has an FPGA add-in module available to target and program FPGA hardware. To simplify the design of complex systems in FPGAs, there exist libraries of predefined complex functions and circuits that have been tested and optimized to speed up the design process. These predefined circuits are commonly called IP cores, and are available from FPGA vendors and third-party IP suppliers (rarely free, and typically released under proprietary licenses). Other predefined circuits are available from developer communities such as Open Cores (typically released under free and open source licenses such as the GPL, BSD or similar license), and other sources [25]. In a typical design flow, an FPGA application developer will simulate the design at multiple stages throughout the design process. Initially the RTL description in VHDL or Verilog is simulated by creating test benches to simulate the system and observe results. Then, after the synthesis engine has mapped the design to a netlist, the netlist is translated to a gate level description where simulation is repeated to confirm the synthesis proceeded without errors. Finally the design is laid out in the FPGA at which point propagation delays can be added and the simulation run again with these values back-annotated onto the netlist.

D. Cross Channel Suppression

The cochlea is the primary sensory organ for hearing. Its three major signal-processing functions are (1) frequency analysis, (2) dynamic-range compression (DRC), and (3) amplification. The cochlea implements these functions in a concurrent manner that does not allow completely separate characterization of each function. Common forms of hearing loss are manifestations of mutual impairment of these major signal-processing functions. Whenever hearing loss includes the loss of DRC, the application of simple, linear gain in an external hearing aid will not restore normal loudness perception of acoustic signals, which is why nonlinear hearing aids with wide dynamic-range compression are sometimes used to ameliorate this problem. An important by-product of DRC is suppression, which contributes to psychophysical simultaneous masking. Two-tone suppression is a non-linear property of healthy cochleae in which the response (e.g., basilar-membrane displacement and/or neural-firing rate) to a particular frequency is reduced by the simultaneous presence of a second tone at a different frequency. Because these invasive measurements cannot be made in humans, suppression must be estimated by other physiological or psychophysical procedures . Distortion product otoacoustic emission (DPOAE) suppression is one of these procedures and can be used to provide a description of the specific suppressive effect of one frequency on another frequency Suppression plays an important role in the coding of speech and other complex stimuli ,and it has been suggested to result in enhancement of spectral contrast of complex sounds, such as vowels .This enhancement of spectral contrasts may improve speech perception in the presence of background noise, although only small improvements have been demonstrated thus far . Sensory hearing loss is defined by elevated threshold due to disruption of the major signal-processing functions of the cochlea. Two-tone suppression is usually reduced in ears with sensory hearing loss .These ears typically also present with loudness recruitment, a phenomenon where the rate of growth of loudness with increasing sound level is more rapid than normal [26]. Multiband DRC hearing-aids attempt to restore DRC but currently do not attempt to restore normal suppression. DRC alone (i.e., without suppression) may reduce spectral contrasts by reducing gain for spectral peaks while providing greater gain for spectral troughs. This paper describes a hearing-aid signal-processing strategy that aims to restore normal cochlear two-tone suppression, with the expectation that this would improve spectral contrasts for signals such as speech. The implementation of suppression in this strategy was inspired by measurements of DPOAE suppression tuning curves (STC). The processes of DRC, amplification and suppression are not implemented separately in this strategy, but are unified into a single operation. The prescription of amplification for the method is based on measurements of categorical loudness scaling (CLS) for tones ,and is intended to restore normal growth of loudness for any type of signal. The strategy is computationally efficient and could be implemented with current hearing-aid technology to restore both normal suppression and normal loudness growth. Restoration of normal suppression may lead to increased hearing-aid user satisfaction and possibly improved speech perception in the presence of background noise. Restoring individualized loudness growth, on the other hand, may increase the usable dynamic range for persons with sensory hearing loss. DPOAE STCs of Gorga et al. provide a comprehensive description of the specific suppressive effect of one frequency on another frequency. These measurements, which are the basis for the suppression component of the signal-processing strategy, are described in Section II. Use of DPOAE-STC measurements in the calculation of gain allows for the implementation of two-tone suppression. Al-Salim et al. described a method to determine the level-dependent gain that a hearing-impaired (HI) listener needs in order to have the same loudness percept for tones as a normal-hearing (NH) listener. These data, based on CLS, provide the basis of our amplification-prescription strategy. Specifically, our strategy aims to provide frequency- and level-dependent gain to a HI listener such that a sound that is perceived as "very soft" by a NH listener is also perceived as "very soft" by a HI listener, and a sound that is perceived as "very loud" by a NH listener is also perceived as "very loud" by a HI listener. The idea is to maximize audibility for low-level sounds while at the same time avoiding loudness discomfort at high levels. The hearing-aid fitting strategy requires CLS data at several frequencies for each HI listener. In order to quantify the deviation from normal, average CLS data for NH listeners is also required. Providing frequency- and level-dependent gain allows for the relative loudness of individual frequency components of complex sounds like speech to be preserved after amplification. The goal of restoring normal loudness growth using actual measurements of loudness is in contrast to current hearing aid fitting strategies Current strategies aim to restore audibility and maximize intelligibility, and primarily use hearing thresholds, average population data and generalized models of loudness, instead of individual loudness data, to prescribe gain. While there have been attempts to use individual measurements of loudness and speech intelligibility, these efforts have been used mainly for refining the fit. Another difference between our proposed fitting strategy and most current fitting methods is that our strategy does not use audiometric thresholds, a model of loudness, or a model of speech intelligibility, but uses instead actual CLS data from each HI individual[27]. This paper describes a novel signal-processing strategy that is motivated by the goals of restoring normal cochlear two-tone suppression and normal loudness growth. The strategy uses a filter-bank to decompose an input signal into multiple channels and a model of two-tone suppression to apply time-varying gain to the output of each channel before subsequently summing these channel outputs. The time-varying gain is designed to implement DRC, amplification and suppression that mimics the way the cochlea performs these functions. These processes are not implemented separately, but concurrently in a unified operation. The gain applied to a particular channel is a function of the levels of all the filter-bank channels not just that channel. The gains are determined by formulas based on (1) the DPOAE-STC measurements of Gorga et al. and (2) the CLS measurements of Al-Salim et al. Suppressive effects are applied to these gains instantaneously because measurements of the temporal features of suppression suggest that suppression is essentially instantaneous. Previous studies have suggested that instantaneous compression has deleterious effects on perceived sound quality. In contrast, pilot data from our laboratory have not identified adverse effects on perceived sound quality due to our implementation of instantaneous compression. However, additional data involving more formal and extensive listening tests will be required to corroborate this preliminary observation.

E. ADC

An analog-to-digital converter (ADC, A/D, or A to D) is a device that converts a continuous physical quantity (usually voltage) to a digital number that represents the quantity's amplitude. The conversion involves quantization of the input, so it necessarily introduces a small amount of error. Instead of doing a single conversion, an ADC often performs the conversions ("samples" the input) periodically. The result is a sequence of digital values that have been converted from a continuous-time and continuous-amplitude analog signal to a discrete-time and discrete-amplitude digital signal.

An ADC is defined by its bandwidth (the range of frequencies it can measure) and its signal to noise ratio (how accurately it can measure a signal relative to the noise it introduces). The actual bandwidth of an ADC is characterized primarily by its sampling rate, and to a lesser extent by how it handles errors such as aliasing. The dynamic range of an ADC is influenced by many factors, including the resolution (the number of output levels it can quantize a signal to), linearity and accuracy (how well the quantization levels match the true analog signal) and jitter (small timing errors that introduce additional noise).

The dynamic range of an ADC is often summarized in terms of its effective number of bits (ENOB), the number of bits of each measure it returns that are on average not noise. An ideal ADC has an ENOB equal to its resolution. ADCs are chosen to match the bandwidth and required signal to noise ratio of the signal to be quantized. If an ADC operates at a sampling rate greater than twice the bandwidth of the signal, then perfect reconstruction is possible given an ideal ADC and neglecting quantization error. The presence of quantization error limits the dynamic range of even an ideal ADC, however, if the dynamic range of the ADC exceeds that of the input signal, its effects may be neglected resulting in an essentially perfect digital representation of the input signal.

An ADC may also provide an isolated measurement such as an electronic device that converts an input analog voltage or current to a digital number proportional to the magnitude of the voltage or current. However, some non-electronic or only partially electronic devices, such as rotary encoders, can also be considered ADCs. The digital output may use different coding schemes. Typically the digital output will be a two's complement binary number that is proportional to the input, but there are other possibilities. An encoder, for example, might output a Gray code. Analog-to-digital conversion is an electronic process in which a continuously variable (analog) signal is changed, without altering its essential content, into a multi-level (digital) signal.

The input to an analog-to-digital converter (ADC) consists of a voltage that varies among a theoretically infinite number of values. Examples are sine waves, the waveforms representing human speech, and the signals from a conventional television camera. The output of the ADC, in contrast, has defined levels or states. The number of states is almost always a power of two – that is, 2, 4, 8, 16, etc. The simplest digital signals have only two states, and are called binary. All whole numbers can be represented in binary form as strings of ones and zeros. Digital signals propagate more efficiently analog signals, largely because digital impulses, which are well-defined and orderly, are easier for electronic circuits to distinguish from noise, which is chaotic. This is the chief advantage of digital modes in communications. Computers ”talk” and ”think” in terms of binary digital data; while a microprocessor can analyze analog data, it must be converted into digital form for the computer to make sense of it. A typical telephone modem makes use of an ADC to convert the incoming audio from a twisted-pair line into signals the computer can understand. In a digital signal processing system, an ADC is required if the signal input is analog.

F. DAC

In electronics, a digital-to-analog converter (DAC, D/A, D2A or D-to-A) is a function that converts digital data (usually binary) into an analog signal (current, voltage, or electric charge). Unlike analog signals, digital data can be transmitted, manipulated, and stored without degradation, albeit with more complex equipment. But a DAC is needed to convert the digital signal to analog to drive an earphone or loudspeaker amplifier in order to produce sound (analog air pressure waves). DACs and their inverse, ADCs, are part of an enabling technology that has contributed greatly to the digital revolution. To illustrate, consider a typical long-distance telephone call. The caller’s voice is converted into an analog electrical signal by a microphone, then the analog signal is converted to a digital stream by an ADC. The digital stream is then divided into packets where it may be mixed with other digital data, not necessarily audio. The digital packets are then sent to the destination, but each packet may take a completely different route and may not even arrive at the destination in the correct time order. The digital voice data is then extracted from the packets and assembled into a digital data stream. A DAC converts this into an analog electrical signal, which drives an audio amplifier, which in turn drives a loudspeaker, which finally produces sound. There are several DAC architectures; the suitability of a DAC for a particular application is determined by six main parameters: physical size, power consumption, resolution, speed, accuracy, cost. Due to the complexity and the need for precisely matched components, all but the most specialist DACs are implemented as integrated circuits (ICs). Digital-to-analog conversion can degrade a signal, so a DAC should be specified that has insignificant errors in terms of the application.

DACs are commonly used in music players to convert digital data streams into analog audio signals. They are also used in televisions and mobile phones to convert digital video data into analog video signals which connect to the screen drivers to display monochrome or colour images. These two applications use DACs at opposite ends of the speed/resolution trade-off. The audio DAC is a low speed high resolution type while the video DAC is a high speed low to medium resolution type. Discrete DACs would typically be extremely high speed low resolution power hungry types, as used in military radar systems. Very high speed test equipment, especially sampling oscilloscopes, may also use discrete DACs.

Digital-to-analog conversion is a process in which signals having a few (usually two) defined levels or states (digital) are converted into signals having a theoretically infinite number of states (analog). A common example is the processing, by a modem, of computer data into audio-frequency (AF) tones that can be transmitted over a twisted pair telephone line. The circuit that performs this function is a digital-to-analog converter (DAC). Basically, digital-to-analog conversion is the opposite of analog-to-digital conversion. In most cases, if an analog-to-digital converter (ADC) is placed in a communications circuit after a DAC, the digital signal output is identical to the digital signal input. Also, in most instances when a DAC is placed after an ADC, the analog signal output is identical to the analog signal input. Binary digital impulses, all by themselves, appear as long strings of ones and zeros, and have no apparent meaning to a human observer. But when a DAC is used to decode the binary digital signals, meaningful output appears. This might be a voice, a picture, a musical tune, or mechanical motion. Both the DAC and the ADC are of significance in some applications of digital signal processing. The intelligibility or fidelity of an analog signal can often be improved by converting the analog input to digital form using an ADC, then clarifying the digital signal, and finally converting the ”cleaned-up” digital impulses back to analog from using a DAC

G. Applicaions in Audio Processing

Most modern audio signals are stored in digital form (for example MP3s and CDs) and in order to be heard through speakers they must be converted into an analog signal. DACs are therefore found in CD players, digital music players, and PC sound cards. Specialist standalone DACs can also be found in high-end hi-fi systems. These normally take the digital output of a compatible CD player or dedicated transport (which is basically a CD player with no internal DAC) and convert the signal into an analog

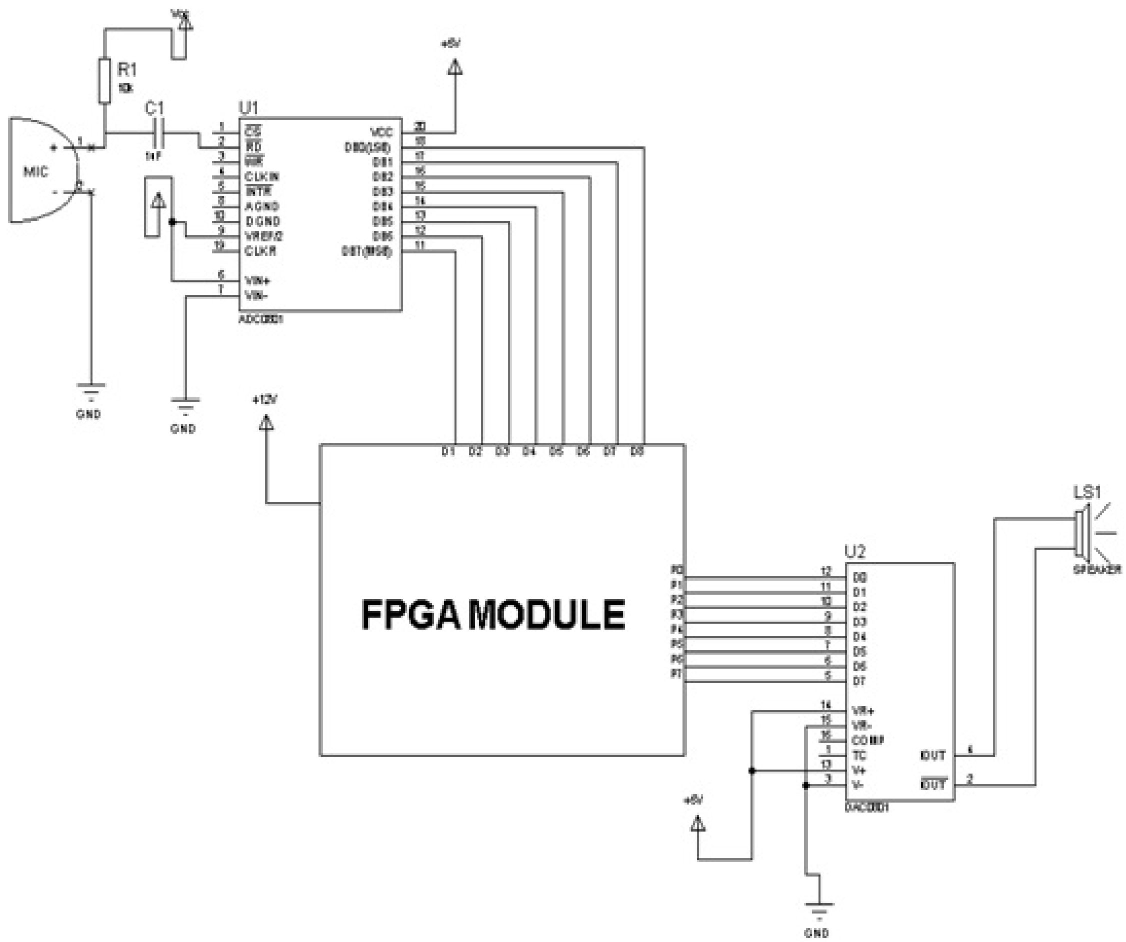

Fig. 2. Hardware schematic of the proposed system.

line-level output that can then be fed into an amplifier to drive speakers [24]. Similar digital-to-analog converters can be found in digital speakers such as USB speakers, and in sound cards. In VoIP (Voice over IP) applications, the source must first be digitized for transmission, so it undergoes conversion via an analog-to-digital converter, and is then reconstructed into analog using a DAC on the receiving party's end

H. Filter Banks

In signal processing, a filter bank is an array of band-pass filters that separates the input signal into multiple components, each one carrying a single frequency sub-band of the original signal. One application of a filter bank is a graphic equalizer, which can attenuate the components differently and recombine them into a modified version of the original signal. The process of decomposition performed by the filter bank is called analysis (meaning analysis of the signal in terms of its components in each sub-band); the output of analysis is referred to as a sub band signal with as many sub bands as there are filters in the filter bank. The reconstruction process is called synthesis, meaning reconstitution of a complete signal resulting from the filtering process.

In digital signal processing, the term filter bank is also commonly applied to a bank of receivers. The difference is that receivers also down-convert the sub bands to a low centre frequency that can be re-sampled at a reduced rate. The same result can sometimes be achieved by under sampling the band pass sub bands. Another application of filter banks is signal compression, when some frequencies are more important than others. After decomposition, the important frequencies can be coded with a fine resolution. Small differences at these frequencies are significant and a coding scheme that preserves these differences must be used. On the other hand, less important frequencies do not have to be exact. A coarser coding scheme can be used, even though some of the finer (but less important) details will be lost in the coding. The vectored uses a filter bank to determine the amplitude information of the sub bands of a modulator signal (such as a voice) and uses them to control the amplitude of the sub bands of a carrier signal (such as the output of a guitar or synthesizer), thus imposing the dynamic characteristics of the modulator on the carrier [28].

The sound signal is received by the microphone the received signal will be in the analog form but the FPGA will receive only digital data so we are using ADC circuit here for converting the analog signal into digital data. In FPGA back ground suppression is done in FPGA after that the digital signal is given to the speaker by converting the digital signal into analog signal, the output will be given through the speaker.

III. Conclusion

In this work, we mainly dealt with cross-channel suppression and amplification of the speech signal for hearing impaired listeners. By using the cross-channel suppression the voice signal is clearly separated from other noises with the help of FPGA. After separating the signal the amplitude range is increased to required level. Then this signal given through head phones to the person.

References

[1] B. C. J. Moore and D. A. Vickers, “The role of spread excitation and suppression in simultaneous masking,” J. Acoust. Soc. Amer., vol. 102, no. 4, pp. 2284–2290, Oct. 1997.

[2] I. Yasin and C. J. Plack, “The role of suppression in the upward spread of masking,” J. Assoc. Res. Otolaryngol., vol. 6, no. 4, pp. 368–377, 2005.

[3] J. Rodr ́ıguez et al., “The role of suppression in psychophysical tone-ontone masking,” J. Acoust. Soc. Amer., vol. 127, no. 1, pp. 361–369, 2009.

[4] M. A. Ruggero, L. Robles, and N. C. Rich, “Two-tone suppression in the basilar membrane of the cochlea: Mechanical basis of auditory-nerve rate suppression,” J. Neurophysiol., vol. 68, pp. 1087–1099, 1992.

[5] W. S. Rhode and N. P. Cooper, “Two-tone suppression and distortion production on the basilar membrane in the hook region of cat and guinea pig cochleae,” Hearing Res., vol. 66, no. 1, pp. 31–45, Mar. 1993.

[6] B. Delgutte, “Two-tone rate suppression in auditory-nerve fibers: Dependence on suppressor frequency and level,” Hearing Res., vol. 49, no. 1–3, pp. 225–246, Nov. 1990.

[7] M. B. Sachs and N. Y. S. Kiang, “Two-tone inhibition in auditory-nerve fibers,” J. Acoust. Soc. Amer., vol. 43, no. 5, pp. 1120–1128, 1968.

[8] H. Duifhuis, “Level effects in psychophysical two-tone suppression,” J.Acoust. Soc. Amer., vol. 67, no. 3, pp. 914–927, 1980.

[9] D. L. Weber, “Do off-frequency simultaneous maskers suppress the signal?” J. Acoust. Soc. Amer., vol. 73, no. 3, pp. 887–893, 1983.

[10] C. Abdala, “A developmental study of distortion product otoacoustic emission (2f1-f2) suppression in humans,” Hearing Res., vol. 121, pp. 125–138, Jul. 1998.

[11] M. P. Gorga, S. T. Neely, D. Konrad-Martin, and P. A. Dorn, “The use of distortion product otoacoustic emission suppression as an estimate of response growth,” J. Acoust. Soc. Amer., vol. 111, no. 1, pp. 271–284, 2002.

[12] M. P. Gorga, S. T. Neely, J. Kopun, and H. Tan, “Growth of suppression in humans based on distortion-product otoacoustic emission measurements,” J. Acoust. Soc. Amer., vol. 129, no. 2, pp. 801–816, 2011.

[13] M. B. Sachs and E. D. Young, “Effects of nonlinearities on speech encoding in the auditory nerve,” J. Acoust. Soc. Amer., vol. 68, no. 3, pp. 858– 875, Mar. 1980.

[14] Ravi K. Kolagotla, Jose Fridman, Bradley C. Aldrich, Marc M. Hoffman, William C. Anderson, Michael S. Allen, David B. Witt, Randy R. Dunton, And Lawrence A. Booth, Jr., “High Performance Dual-Mac DSP Architecture”, IEEE Signal Processing Magazine, July, 2002, pp. 42-53.

[15] James M. Kates, “Feedback Cancellation in Hearing Aids Using Constrained Adaptation” , Workshop on Applications of Signal Processing to Audio and Acoustics, New Pulfz, New York, Oct. 17-20, 1999.

[16] Neeraj Magotra Juan G. Vargas-Rubio, “Comparison Of Two Open-Loop Adaptive Speech Enhancement Algorithms for Digit Al Hearing Aids”, IS-CAS 2000 - IEEE International Symposium on Circuits and Systems, May 28-31, 2000, Geneva, Switzerland.

[17] Marcio G. Siqueira, Abeer Alwan, “Steady-State Analysis of Continuous Ad-aptation in Acoustic Feedback Reduction Systems for Hearing-Aids” IEEE Transactions on Speech and Audio Processing, Vol. 8, No. 4, July 2000 pp. 443-453.

[18] Boaz Rafaely and Mariano Roccasalva-Firenze, “Control of Feedback in Hear-ing Aids—A Robust Filter Design Approach”, IEEE Transactions On Speech And Audio Processing, Vol. 8, No. 6, pp. 754-756, November 2000.

[19] Johan Hellgren and Urban Forssell, ” Bias of Feedback Cancellation Algo-rithms in Hearing Aids Based on Direct Closed Loop Identification”, IEEE Transactions on Speech and Audio Processing, Vol. 9, No. 7, pp. 906-913, November 2001.

[20] Stefan Gustafsson, Rainer Martin, Peter Jax, and Peter Vary, “ A Psycho-acoustic Approach To Combined Acoustic Echo Cancellation And Noise Reduction”, Ieee Transactions On Speech And Audio Processing, Vol. 10, No. 5, July 2002, pp. 245- 256.

[21] Yong Lia, and Ying Wei “A Computationally Efficient Nonuniform FIR Digi-tal Filter Bank For Hearing Aids”, IEEE Transaction on Circuits and Systems, Vol. 52, No. 12, December, 2005, pp. 2754 – 2762.

[22] Thomas Fillon and Jacques Prado, “Acoustic Feedback Cancellation For Hear-ing-Aids, Using Multi-Delay Filter”, 5th Nordic Signal Processing Symposium, October 4-7, 2002, on board Hurtigruten, Norway.

[23] Johan Hellgren "Analysis of Feedback Cancellation in Hearing Aids With Fil-tered-X LMS and the Direct Method of Closed Loop Identification", IEEE Transactions on Speech and Audio Processing, Vol. 10, No. 2, pp. 119-131, February, 2002.

[24] Tomas Gänsler and Jacob Benesty, "New Insights Into the Stereophonic Acoustic Echo Cancellation Problem and an Adaptive Nonlinearity Solution", IEEE Transactions On Speech And Audio Processing, Vol. 10, No. 5, July 2002 257, pp. 257-267.

[25] Young-cheol Park, In-young Kim and Sang-min Lee "An efficient adaptive feedback cancellation for hearing aids", Proceedings of the 25th Annual International Conference of the IEEE on Engineering in Medicine and Biology Society, 2003, Vol. 2, pp. 1647-1650.

[26] Hsiang-Feng Chi, Shawn X. Gao, Sigfrid D. Soli and Abeer Alwan "Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids", Journal of Speech Communication, Elsevier, Vol. 39, Issue -2, January 2003, pp. 147-161.

[27] Ann Spriet, Marc Moonen and Jan Wouters "Stochastic Gradient-Based Im-plementation of Spatially Preprocessed Speech Distortion Weighted Multichannel Wiener Filtering for Noise Reduction in Hearing Aids " IEEE Transactions on Signal Processing, Vol. 53, No. 3, March 2005 911, pp. 911-925.

[28] Ngwa Abinwi Shusina and Boaz Rafaely "Unbiased Adaptive Feedback Can-cellation in Hearing Aids by Closed-Loop Identification", IEEE Transactions On Audio, Speech, And Language Processing, Vol. 14, No. 2, March 2006, pp. 658-665.