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Hear-Clear

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Abstract— **People with hearing loss problem use hearing aids, which are nothing but a audio amplifier. Analog hearing aids are less flexible and are simply designed to gather sound energy and direct it into the ear canal at fixed volumes for all the frequencies of sound. But with the availability of miniature microphone sensors and extremely low power digital signal processors, present day digital hearing aids can be constructed with audiometrical and cognitive intelligence that matches the hearing loss, physical features and lifestyle of the wearer, thus delivering more value and benefits for the hearing impaired person.The objective of the paper is to design a smart earplug system integrated with non-invasive bone conduction technique which is capable of doing advanced audio processing to provide voice enhanced, noise filtered audio for the hearing impaired people. The developed system in this work is compatible to work as an embedded music player, a life activity tracker and a Smartphone companion. It can even translate the SMS that is just received on your Smartphone into voice and feed to ear.**

Keywords—Bone conduction, earplug, microphone, digital signal processors.

I.INTRODUCTION

Auditory perception is one of the most important senses of human With a focus on sound propagation, auditory perception can be categorized mainly into two: air-conduction (AC) and Bone conduction (BC) sound. AC sound propagates from outer ear to the inner ear via middle ear, and the dominant component of the auditory perception from the external environment. Against this normal pathway, BC sound propagates through the skin and skull. Previously, as the BC sound does not require middle ear functionality, it is widely used for hearing aid. Air Conduction is the transmission of sound vibrations to the eardrum through the external auditory mellitus. Transmission of sound to the inner ear is through the external auditory canal and middle ear. Sound that travels through the outer ear impacts on the eardrum, and causes it to vibrate. The three ossicles bones transmit this sound to a second window (the oval window) which protects the fluid-filled inner ear. In detail, the pinna of the outer ear helps to focus a sound, which impacts on the eardrum. The malleus rests on the membrane, and receives the vibration. This vibration is transmitted along the incus and stapes to the oval window. Two small muscles, the tensor tympanium and stapedius, also help modulate noise. Bone conduction is the conduction of sound to the inner ear through the bones of the skull. Bone conduction transmission can be used with individuals with normal or impaired hearing. Bone conduction is one reason why a person's voice sounds different to them when it is recorded and played back. Because the skull conducts lower frequencies better than air, people perceive their own voices to be lower and fuller than others do, and a recording of one's own voice frequently sounds higher than one expects it to sound. Although hearing requires an intact and functioning auditory portion of the central nervous system as well as a working ear, human deafness (extreme insensitivity to sound) most commonly occurs because of abnormalities of the inner ear, rather than in the nerves or tracts of the central auditory system.

Kristian Timm Andersen et al suggested an adaptive time-frequency analysis scheme along with a synthesis scheme using an asymmetric window [1]. The proposed scheme is suitable for audio noise reduction with a low delay in the range of 0 to 4ms. The main novelty of the paper is the adaptive analysis scheme that can adapt to the incoming signal independently in both time and frequency by employing a complex filter on a DFT modulated filter bank. Noise reduction task is also performed indicates a good performance compared to reference implementations in terms of segmental SNR and PESQ. J. Agnew and J. M. Thornton proposed a real-time audio device to present a clear and audible signal to the user at a low delay [2]. For a hearing aid in particular, a low delay is critical since sound traveling through the vent into the ear canal should not get too out of sync with the sound coming from the hearing aid speaker. Studies have shown that delays exceeding approximately 10ms can be objectionable while delays around 3-5ms can still be detected. This delay includes buffering and would also include A/D and D/A conversion. In this paper they consider low delay to mean a filter bank that can apply a frequency dependent gain to the signal in around 4ms or less. This delay is not a hard limit, since the total delay also includes buffering and computational delay, although these delays should be small in comparison to ensure a truly low delay implementation.

H. W. Lollmann and P. Vary proposed to enhance the sound quality using noise reduction, which in the single-channel case is done by applying a frequency-dependent gain to the signal [3]. To achieve some tradeoff between time and frequency resolution in the noise reduction, the filter bank is often designed to have a non-uniform frequency resolution with

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more narrow bands in the low frequencies. A popular framework for non-uniform frequency resolution is the wavelet transform, for instance the critically sampled tree-structured filter bank. For our application, however, the iterated use of the so-called "mother" wavelet results in a high group delay that makes it inappropriate for very low delay applications. Also, the need for noise reduction in the filter bank necessitates oversampling to avoid aliasing in the reconstruction. R. Hendriks et al suggested an adaptive time-frequency (TF) resolution schemes. Also, it has been shown that adapting the TF resolution can lead to improvements in noise reduction. Common for these approaches is that they are not suitable for low delay implementations since they require longer time windows to determine the TF resolution and/or that they have a high computational complexity. A window switching approach to adaptive TF resolution with a low delay of 10ms has been proposed in.

Falco Strasser and Henning Puder proposed a sub-band feedback cancellation system which combines decorrelation methods with a new realization of a known non-parametric variable step size. To apply this step size in the context of adaptive feedback cancellation, a method to estimate the signal power of the desired input signal, i.e., without feedback, is necessary. The complete system is evaluated extensively for several speech and music signals as well as for different feedback scenarios in simulations with feedback paths measured in concrete applications as well as for real-time simulations with hearing aid dummies. Both use hearing loss compensation methods as applied in physical hearing aids. The performance is measured in terms of being able to prevent entrainment and to react to feedback path changes. For both simulation setups the system shows a good performance with respect to the two performance measures. Furthermore, the overall feedback cancellation method relies only on few parameters, shows a low computational complexity, and therefore has a strong practical relevance.

In the existing scenario, the hearing aids perform amplification to the air conduction signal alone and the bone conduction signal is not considered. The air conduction signal is affected by many reasons and some cases it cannot be fully recovered. In our day to day life we have seen many hearing impaired people using hearing aids. Most of these hearing aids use the concept of air conduction. In the proposed system a hearing aid which uses bone conduction technique. We have added additional features to the aid that it can be interfaced with the smart phone and also used as an activity tracker.

II MATERIAL & METHODS

The development of the system could be understood from the block diagram shown in figure 1. In the proposed system, the hearing aids are to amplify the bone conduction signal.

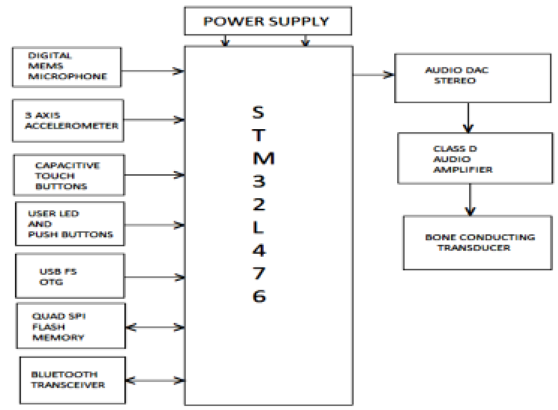

Fig 1. Block diagram of the proposed system

With the availability of miniature microphone sensors and extremely low power digital signal processors, hearing aids can be constructed with audio metrical and cognitive intelligence that matches the hearing loss, physical features and lifestyle of the wearer, thus delivering more value and benefit for the hearing impaired person [5].

2.1 STM32L476

The STM32L476 device shown in fig.2 is an ultra-low-power microcontrollers based on the high-performance ARM® Cortex®-M4 32-bit RISC core operating at a frequency of up to 80 MHz. The Cortex-M4 core features a Floating point unit (FPU) single precision which supports all ARM single-precision data-processing instructions and data types. It also implements a full set of DSP instructions and a memory protection unit (MPU) which enhances application security.

Fig 2. STM32L476

2.2 MP34DB01 MEMS Microphone

The MP34DB01 shown in the Fig. 3 is an ultra-compact, low-power, omni directional, digital MEMS microphone built with a capacitive sensing element and an IC interface with stereo operation capability. The sensing element, capable of detecting acoustic waves, is manufactured using a specialized silicon micromachining process dedicated to produce audio sensors. The IC interface is manufactured using a CMOS process that allows designing a dedicated circuit able to provide a digital signal externally in PDM format. The MP34DB01 has an acoustic overload point of 120 dBSPL with a best on the market 62.6 dB signal-to-noise ratio and -26 dBFS sensitivity. The MP34DB01 is available in a

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bottom-port, SMD-compliant, EMI-shielded package and is guaranteed to operate over an extended temperature range from -40 °C to +85 °C.

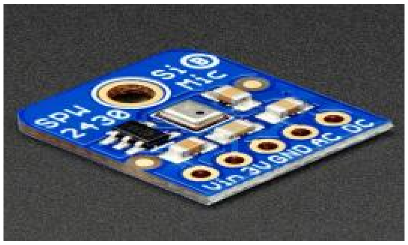

Fig 3. MEMS Microphone

2.3 Bluetooth Transceiver

HC-05 module shown in the Fig. 4 is an easy to use Bluetooth SPP (Serial Port Protocol) module, designed for transparent wireless serial connection setup. Serial port Bluetooth module is fully qualified Bluetooth V2.0+EDR (Enhanced Data Rate) 3Mbps Modulation with complete 2.4GHz radio transceiver and baseband. It uses CSR Bluecore 04-External single chip Bluetooth system with CMOS technology and with AFH(Adaptive Frequency Hopping Feature). It has the footprint as small as 12.7mmx27mm. Hope it will simplify your overall design/development cycle.

Fig 4. Bluetooth Transceiver

2.4 USB(Universal Serial Bus)

USB shown in the Fig.5 is started as a standard for connecting peripherals (such as mice, keyboards) to PCs. Now it is the standard for connecting personal device (iPod's, printers) to pc and personal devices (PDA, scanner)to personal devices(printer, iPod's). Microchip offers USB products – Peripherals, Embedded Hosts and OTG in PIC18, PIC24 and PIC32 portfolio.

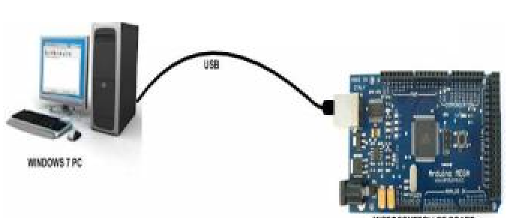

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Fig 5. *USB(Universal Serial Bus)*

2.5 CS43L22 Audio DAC

The CS43L22 is a highly integrated, low power stereo DAC with headphone and Class D speaker amplifiers. The CS43L22 offers many features suitable for low power, portable system applications. The DAC output path includes a digital signal processing engine with various fixed function controls. Tone Control provides bass and treble adjustment of four selectable corner frequencies. Digital Volume controls may be configured to change on soft ramp transitions while the analog controls can be configured to occur on every zero crossing. The DAC also includes de-emphasis, limiting functions and a BEEP generator delivering tones selectable across a range of two full octaves. The stereo headphone amplifier is powered from a separate positive supply and the integrated charge pump provides a negative supply. This allows a ground-centered analog output with a wide signal swing and eliminates the need for external DC-blocking capacitors. The Class D stereo speaker amplifier does not require an external filter and provides the high efficiency amplification required by power sensitive portable applications. The speaker amplifier may be powered directly from a battery while the internal DC supply monitoring and compensation provides a constant gain level as the battery's voltage decays. The CS43L22 accommodates analog routing of the analog input signal directly to the headphone amplifier. This feature is useful in applications that utilize an FM tuner where audio recovered over-the-air must be transmitted to the headphone amplifier directly.

2.6 Capacitive Touch Keypad Sensor

The MPR121 shown in the Fig. 6 is the second generation capacitive touch sensor controller after the initial release of the MPR03x series devices. The MPR121 features increased internal intelligence, some of the major additions include an increased electrode count, a hardware configurable I2C address, an expanded filtering system with debounce, and completely independent electrodes with auto-configuration built in. The device also features a 13th simulated sensing channel dedicated for near proximity detection using the multiplexed sensing inputs.

Fig 6. *Capacitive Touch Keypad Sensor*

2.7 Bone Conducting Transducer

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Bone conduction is the conduction of sound to the inner ear through the bones of the skull. Bone conduction transmission can be used with individuals with normal or impaired hearing. Bone conduction is one reason why a person's voice sounds different to them when it is recorded and played back. Because the skull conducts lower frequencies better than air, people perceive their own voices to be lower and fuller than others do, and a recording of one's own voice frequently sounds higher than one expects it to sound. Musicians may use bone conduction while tuning stringed instruments to a tuning fork. After the fork starts vibrating placing it in the mouth with the stem between the back teeth ensures that one continues to hear the note via bone conduction, and both hands are free to do the tuning. Some hearing aids employ bone conduction, achieving an effect equivalent to hearing directly by means of the ears. A headset is ergonomically positioned on the temple and cheek and the electromechanical transducer, which converts electric signals into mechanical vibrations, sends sound to the internal ear through the cranial bones. Likewise, a microphone can be used to record spoken sounds via bone conduction. The first description, in 1923, of a bone conduction hearing aid was Hugo Gernsback's "Osophone", which he later elaborated on with his "Phonosone". After the discovery of Osseo integration around 1950 and its application to dentistry around 1965, it was noticed that implanted teeth conducted vibrations to the ear. As a result, bone anchored hearing aids were developed and implanted from 1977 on.

2.8 Atollic True STUDIO® IDE

The Atollic True STUDIO® IDE provides a modern and highly integrated development environment which directly supports the use of advanced workflow tools such as version control, bug tracking, code review, code analysis and distributed task-based development, along with tailored control for project and build control and a fully integrated debugger. The Atollic True STUDIO® IDE comes in a variety of packages enabling customers to select the features/price model best suited to their development needs. As the underlying compiler tool chain is based on the GNU C/C++ compiler, there is no worry about a 'proprietary' tool chain becoming out of date, or unavailable. The same goes for the Atollic True STUDIO® IDE, as it is based on the open Eclipse framework. A standard firmware library infrastructure has been created by ARM Ltd. along with semiconductor and tool chain vendors.

The Cortex® Microcontroller Software Interface Standard (CMSIS) defines a hardware abstraction layer which is available as a firmware library coded to support compilation by a number of compilers, including the GNU C/C++ compiler and the IAR Embedded Workbench® C/C++ compiler. Details can be found on the ARM®. The firmware generated by the Atollic True STUDIO® IDE for the ARM® Cortex® series of processors includes all low-level device control via the CMSIS firmware library (including startup, interrupt and exception handlers) along with chip vendor supplied peripheral device drivers. As the firmware library compile to a standard, and has been written to support both the GNU and IAR Embedded Workbench® compilers (by using conditional compilation), users should find that they have a familiar Application Programming Interface (API) to code against, which reduces the porting exercise to one of tuning the build control and porting application source files.

2.9 Functional Description

The block diagram of the proposed hearing aid is shown in Fig.1. It consists of STM32L476 which is a digital signal controller, MEMS microphone for converting audio signal into digital signal, DAC for converting processed digital signal from STM32L476 into analog signal, Class D audio amplifier for audio amplification, bone conducting transducer for converting digital signal into mechanical vibrations, capacitive touch keypad for changing applications, Quad SPI flash memory for storing audio files, Bluetooth transceiver for interfacing with smart phones, accelerometer for activity tracking and USB for interfacing with PC.

The audio signal is given to the MEMS microphone which will convert the audio signal into electrical signal. This electrical signal is given to the digital signal controller STM32L476 for further processing. [6],[7] Then signal is amplified and given to the bone conducting transducer which will produce vibrations so that human can hear.

The main component of this hearing aid is the STM32L476 which is a microcontroller (MCU) features a ARM cortex M4F core. It has its own MEMS based microphone. It has two applications. One is used for record application and other is for player application.

Record application uses the MP34DT01 MEMS digital microphone to store 16-bit audio samples at 48kHz.The recorded audio file is stored in the QUAD-SPI Flash memory. Note that recording a new audio file will overwrite the previous one. The player application plays back in loops any WAV stored in the internal Flash or external Quad-SPI Flash memory. If both memories contain a WAV file, one can select from a sub-menu which one want to play. It uses a CS43L22 audio code to output the audio data on the 3.5mm jack. Earphone volume can be adjusted during playback.

In this hearing aid the STM32L476 is operated in the record application. Audio from external source is given to MEMS microphone which is embedded in it. Audio from external source can include human voice. Since the chip is in record application it will convert the audio signal into electrical signal. This electrical signal is a digital signal because digital MEMS microphone produces output as digital signal and also digital signal is only needed for the further processing. This conversion is done because STM32L476 is a digital signal controller which process only digital signals [8]. This digital signal is stored in the memory of STM32L476 and also given for further processing. STM32L476 contains FIR filter [9]. The digital signal from MEMS microphone is given to FIR filter. This filter will remove the noises present in it. This is because human ears can able to hear only low frequency signals. To facilitate this factor noise removal is done by the FIR filter which is present in the STM32L476.

The noise removed signal is given to the chip called CS43L22.This chip is embedded with digital to analog converter and also with Class D audio amplifier. Thus the noise removed signal is given to digital to analog converter which will convert the digital signal into analog signal. Due to noise removal by the FIR filter embedded in the STM32L476, signal level goes below the audible level. To overcome this problem amplification is provided. To accomplish the work of amplification, Class D audio amplifier is engaged. As mentioned above Class D stereo amplifier is present in the chip CS43L22. This chip CS43L22 can be externally connected to STM32L476. The analog signal given to the Class D audio amplifier gets amplified and given to the bone conducting transducer.

Bone conducting transducer is another important component in this hearing aid. The main objective of the hearing aid is based on the bone conduction technique. This can be accomplished by this bone conducting transducer only. Bone conducting transducer will convert the given analog signal into mechanical vibrations. This vibration is given to the skull bone of the human being so that human can hear. These mechanical vibrations can be given to the bones of skull by placing the bone conducting transducer on the skull.

The additional features of this hearing aid includes music player, smart phone companion, SMS reader, activity tracker and can be interfaced with PC. As said earlier STM32L476 is composed of external QUAD-SPI flash memory. In this memory, audio files can be stored so songs are stored. Operating the STM32L476 in the player application one can hear the music through bone conducting transducer. But the songs which are stored in the flash memory can only be played. The hearing aid can be interfaced with the smart phone using Bluetooth. Using Bluetooth transceiver HC-05, this hearing aid can be interfaced with smart phone. Since it is interfaced with smart phone, this hearing aid can read the SMS that is received by the smart phone. But the SMS read by the hearing aid is predefined. [4] discussed about a system, GSM based AMR has low infrastructure cost and it reduces man power. The system is fully automatic, hence the probability of error is reduced. The data is highly secured and it not only solve the problem of traditional meter reading system but also provides additional features such as power disconnection, reconnection and the concept of power management.

The hearing aid can also be used as activity tracker. The digital signal controller STM32L476 contains accelerometer embedded in it. Using this accelerometer, activity of the person who is using the hearing aid can be tracked. This accelerometer gives values in three directions and is given to the smart phone using Bluetooth. The hearing aid can also connect with PC by using USB (Universal Serial Bus). The hearing aid contains multiple features like hearing live audio, music player, SMS reader and activity tracker. To change these applications according to the user's wish capacitive touch keypad is used. Using the capacitive touch keypad user can change the modes of the hearing aid. Hearing aid can also be used for different applications as mentioned above, by controlling it with smart phone interfaced by the Bluetooth.

III. RESULTS & DISCUSSION

The proposed hearing aid contains multiple applications. In order to change the applications control is needed. This control is provided by capacitive touch keypad controller as well as by the smart phone using Bluetooth. The screen shown in figure 7 indicates different applications so that user can change according to their requirement.

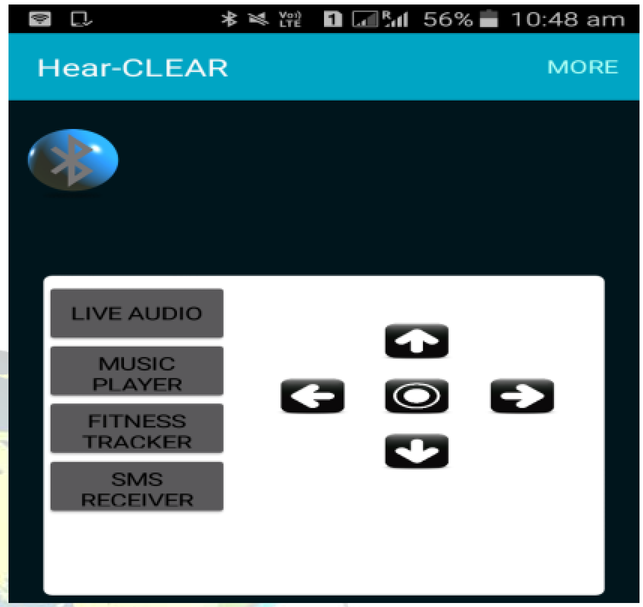

Fig. 7 Screen for Changing Applications Using Smartphone

In this hearing aid bone conducting transducer is placed on the skull of the human being. (Eg. bone behind the ear) as shown in the Fig 7. The hearing aid contains additional features like music player, SMS reader, activity tracker and these features has been successfully implemented in hardware and demonstrated with different persons. Since the hearing aid contains multiple features, this application can be changed by using capacitive touch keypad and smart phone.

Fig 7. *Bone transducer placed on the ear bone*

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The proposed hearing aid is capable of converting the SMS received by the smart phone into text and play it. The person by changing the application can be able to hear the SMS that is received by the smart phone. The SMS read by the hearing aid is predefined one. So in future it can be modified to read all SMS received by the smart phone.

IV. CONCLUSION

The proposed hearing aid amplifies the signal perceived through bone conduction and the same is implemented and tested. The live audio and the music can be heard after proper conversion. The predefined SMS can be heard as voice and the activity can also be tracked using the accelerometer. In future, the hearing aid can be advanced by blending both Air conduction and Bone conduction signals together. Presently the proposed system used the music and the SMS from the standard source. In future, live music and hear the voice contents of the instant SMS delivered to the phone.

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