Software Defined Hearing Aid

Richard Knowles, and David Vaz; Advisor: Dr. Allen Katz
Department of Computer Engineering, The College of New Jersey, Ewing, NJ

Abstract

The long term objective of this project is to design a hearing aid that will assist in both hearing and short-term memory loss. It is hoped that this combination can treat both hearing and memory issues. The principal tasks accomplished this year are the following: 1) create the software for the audio filters needed by the hearing aid system with focus on an adjustable filter that can complement for the hearing loss of individual hearing aid users; and 2) create the software for an automatic level control needed by the hearing aid system. Adjustable level control is essential for the efficient operation of the hearing aid system. The final product shall be a software-defined hearing aid that is very user friendly and treats both short-term memory and hearing loss.

Introduction/Background

15% (26 million) of Americans between the ages of 20 and 69 lose hearing.
Age groups:
- 45-64 years old: 18%
- 65-74 years old: 30%
- 75 years or older: 47%

Modified Design

- Between the microphone and the amplifying phase is a digital system which opens new possibilities that can be done with a hearing aid.
- With the CPU and DSP aspect included, sound can be recorded as well as modified by the user.

Benefits

Improved hearing/short term memory
- A PDA with suitable memory allows sound segments to be recorded
Moving DSP outside system provides more capabilities
- Easier functionality
- Hearing aid can be used strictly as an amplifier
Wireless, utilized Bluetooth technology
Compact, easy to transport
Includes graphic equalizer for user to adjust sound accordingly

Program Design

32 frames per block of audio, 16 bit audio both L and R
Each frame is 4 bytes (2 bytes each channel)
- Each while loop in code is 128 bytes
In order for CD quality audio, system uses 44.1 kHz sampling rate (44,032 used in code)
- ALC and equalizer processing implemented in 11.025 kHz
3 MB, equivalent to 32 seconds, is allocated for memory

Equalizer Program

Requests user to enter the values for 6 gains, one for each band.
Reads in sound input and stores the values in an array
- Counts the number of samples and divides them evenly into 6 sectors.
Each sector gets modified accordingly by the gain values entered by the user using the FIR function

FIR Function

- An array of 161 values (16 taps)
- Values are symmetrical about the middle value, therefore to speed up program, 81 values are used
- Coefficient values are multiplied by audio input values, creating a total of 81 multiplications.
- 81 values are summed up, creating a resultant value of ‘fsum’

FIR Filter

Results

The waveforms above show the filtered outputs of a sine sweep going through the six bands of the equalizer. Output varies depending on the gain set for each band.

Design Tools

Microsoft Windows XP
- XP provides better functionality for our task because of the availability of .wav files
Microsoft Visual Studio/Dev C++
- Used to write and compile C++ code
GW BASIC
- Used for calculation of coefficients in FIR filter

Conclusion

Hearing and memory loss are detrimental to a person’s lifestyle. Although conventional hearing aids can assist those with hearing impairments, they are unable to help those with memory loss. This device contains a digital system that processes sound and delivers it to an ear piece via Bluetooth technology. By processing sound digitally, audio can be stored into memory and called upon command. Using the digital system, a graphic equalizer was successfully implemented into the system in real-time. An ALC is currently in development and should be completed in the near future.

Acknowledgements

Jay Ross
- Technical advisor
Ted Moskalenko
- Programmer of last year’s project