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Real Time Noise Cancellation and Improving Speech Intelligibility of Sensor neural Hearing Impaired by Using DSP Processor

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Abstract: In human body out of the three hearing losses, the foremost extreme listening to loss that we tend to can't do surgery is sensor neural listening to loss. Sensor neural hearing problem is one style of hearing disorder, during which causes stops operating within the receptor. Sensor neural disablement causes reduction within the notion of speech and widening of sense of hearing filters. Widening of sensory system filter leads to central masking. Spectral masking leads to degraded speech belief as there is also covering of frequency parts with the help of adjacent frequency parts. Earlier studies have state that, frequency protective is also reduced by method of sound recorded by electro-acoustic transducer devices that causes dichotic presentation, exploitation necessary band filter institution. For someone with paying attention to impairment the range of levels between weakest sound which will be detected and also the most intense sound which will be tolerated is a smaller amount than everyday auditor. For compensating this, hearing aids expand vulnerable sounds larger than they create larger extreme sound.

Keywords: Sensor neural, numerous, filter.

I. INTRODUCTION

Hearing problems may be categorized, on the premise of vicinity of the disorder within the auditory gadget, as conductive, sensor neural, and imperative losses. Conductive hearing loss happens due to an abnormality inside the center ear leading to negative transmission of the sound to the internal ear. The traits of sensor neural listening to loss, which happens because of the damage of hair cells within the cochlea or degeneration of auditory nerve fiber. The sensor neural hearing impairments famous increase in thresholds of listening to, discount in dynamic range of hearing, degradation of temporary resolution, growth in temporal protecting and decreases frequency due to boom in spectral overlaying. The basic idea is discount in the spectral protecting to increase the working condition of the speech recognition indicators.

In standard, sensor neural loss is hard to treatment and it turns into step by step worse with time. Sensor neural listening to loss is related to widening of the auditory clear out bandwidths for that reason the filter out slope will become shallower and consequences in overlap of adjacent spectral bands known as spectral protecting. Splitting of speech into two alerts, such that the frequency additives which might be in all likelihood to get masked are separated and presented to unique ears has helped in reducing the impact of spectral covering. The fact from the alerts offered to the two ears gets included at better degrees in the auditory system.
The FPGA shape allows parallel and compound facts processing operations every clock cycle, while the DSP-chip primarily based processing entails sequential education fetch-and execute cycles. Because of strength and area constraints, hearing aids are typically designed using ASIC (software specific integrated circuit), which involves sizable nonrecurring price related to chip fabrication. This cost can be averted by way of using FPGA for immediate prototyping earlier than taking the layout to ASIC stage. As filtering may be realized the usage of a parallel structure, an FPGA platform is used for imposing the brush filters for use in binaural hearing aids. After done verification, the FPGA System-based layout can be converted into ASIC layout.
Many sports in human every day keep involve process of audio facts. Tons data close to the setting is obtained through the perceived acoustic sign. in addition a good deal interaction between personalities takes place through audio story, and also the capability to listen and system thousand is important soon take a locality of conversations with totally different individuals. As individuals end up being older, the potential to listen to sounds degrades. Hereby, being attentive to impair unfastened their ability to tune sounds in howling environments and consequently the flexibility follow conversations.

One amongst the targets of being attentive to aids is to boost the intelligibility and thereby assist citizenry to suits high conversations sound. one amongst the ways to boost the speech and to compress the ground noise. Nowadays, exceptional techniques exist so it'll beautify speech, and herewith increase the understandability in howling environments. Speech improvement ways will either be based on one electro-acoustic transducer recording. In

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speech improvement techniques, a desired speech signal is gift in noise, and these signal may be increased the speech sign or by victimization the noise.

II. PROBLEM DEFINITION

i. In widespread sensorineural loss is hard to treatment and it turns into steadily worse with time.
ii. This loss is related to widening of auditory clear out BW for that reason the filter slope will become shallower and consequences in overlap of adjoining spectral bands called spectral protecting. This leads to decrease in frequency resolving potential of auditory machine of ears.
iii. it's far nice to break up the speech into one-of-a-kind bands and supplying the alternate bands to every ear.
iv. For this cause wiener and comb filtering approach is used on the way to split the speech into two signals such that frequency additives that are possibly to get masked are separated and offered to exceptional ears has helped in decreasing the effect of spectral covering.

III. OBJECTIVES

i. To design DSP and Raspberry Pi based totally simulator for processing of speech with the aid of wiener and comb filtering technique.
ii. To improve the speech belief for listeners with excessive frequency listening to loss.
iii. To acquire a reconstructed speech sign which is much like input speech signal.

IV. PROPOSED SYSTEM: DIGITAL HEARING AID

Digital hearing resource procedure the speech sign within the equal way as human ear capabilities. Human ear has three primary element outer ear, center ear and inner ear. Outer ear receives the signal from outdoor world and direct closer to the center ear, in virtual hearing resource directional microphone plays this venture. Middle ear acts as an impedance matching network and as an amplifier. Predominant function which center ear performs is the distribution of frequency bands into numerous small bands.

Virtual paying attention to helpful resource uses some variety of electrical resistance matching network in conjunction with the electronic equipment. sign process algorithms which has DFFT, FFT, Uniform or Non-Uniform filter financial organization is enforced on FPGA to cut up the entire waveband into many little bands so desired band of frequency is manipulated as consistent with the need of affected person.
This presents the power to the doctors to change the hearing parameters of a affected person on the equal being attentive to helpful resource with none various of component whereas standard analogue hearing resource doesn't supply this ability. sense organ capabilities as a spectrum instrument that encodes the sign at various frequency, enlarge the low amplitude sign and compress the signal that is higher than the hearing capability of human, decrease the noise power, makes one amongst a sort nerve cells resonates and within the finish transmits the short pulses to the mind.

In virtual hearing aid those features are finished by using various sign processing algorithms which includes adaptive filtering, comments and echo cancellation, dynamic range compression, and finally the usage of the IFFT algorithm and DAC, frequency area sign converted back into continuous time area and brought to the speaker.

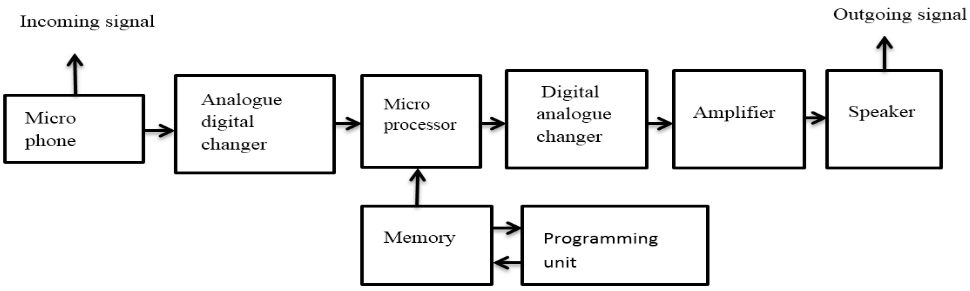

Fig. Structure of Digital Hearing Aid

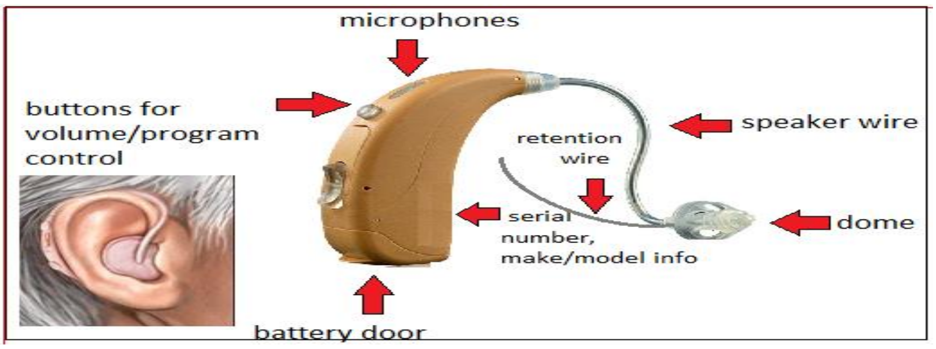

Fig. Structure of Human Ear

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Above figure affords a pictorial evaluation between human ear and virtual hearing useful resource. This method proposes improvement in speech intelligibility for sensor neural listening to impaired. Covering in sensor neural loss is a phenomenon where presences of 1 signal component effects in audibility of the neighboring sign factor. The record received from both the ears receives integrated. Subsequently, splitting of information in speech sign for presenting alerts to the two ears in some form of a complimentary style allows in reducing the impact of protecting.

The multiplied temporal masking outcomes in improved forward and backward masking of vulnerable acoustic segments via strong ones.

V. SYSTEM IMPLEMENTATION

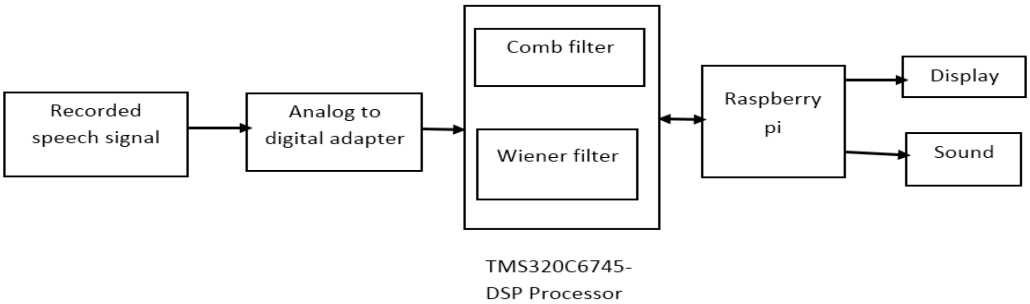

Fig: Block Diagram of System

VI. THE RESULTS FOR NOISE CANCELLATION FOR DIFFERENT AUDIO SIGNALS ON MATLAB AND DS PROCESSOR:

RESULTS ARE BASED ON ITS SIMULATION RESULTS

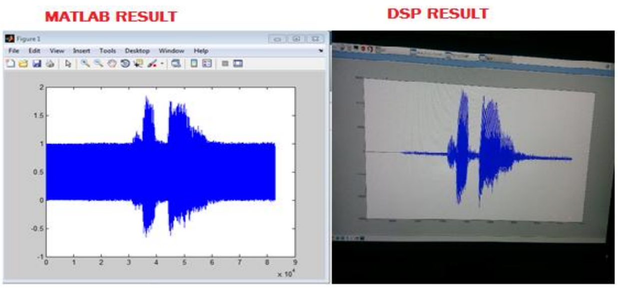

Fig: waveform of clean word" dettol"

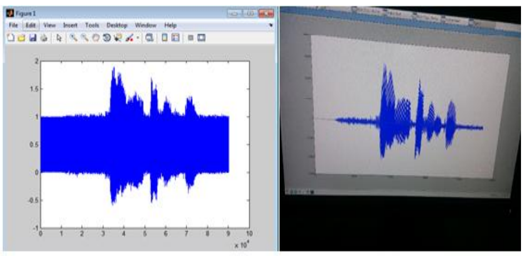

Fig: waveform of clean word" bournvita"

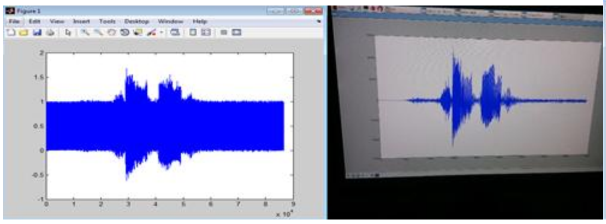

Fig: waveform of clean word" Biscuit"

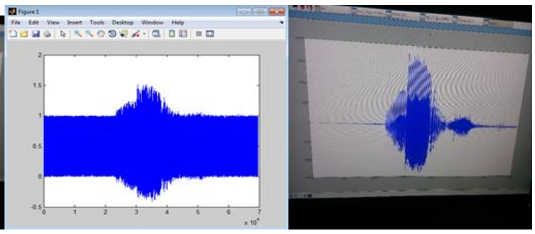

Fig: waveform of clean word" Santoor"

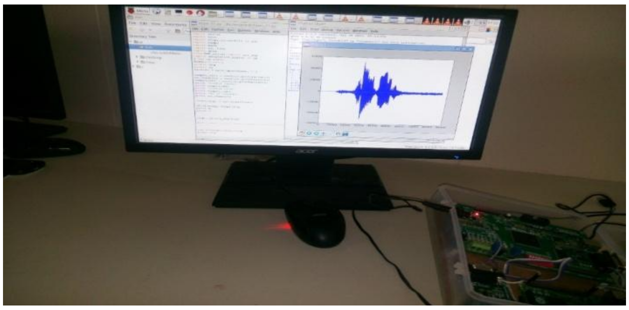

Fig: Test setup of the System

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VII. CONCLUSION

The effects here by conclude that the evolved DSP platform for digital hearing useful resource using Raspberry pi is labored for exclusive audio alerts. The outputs of this device indicates that the unique speech signal that is by some means corrupted with environmental noise is get again through the digital hearing device with suppressed noise signal and with compressed amplitude.
The device additionally indicates that the used of raspberry pi to plays the algorithms for digital paying attention to resource is best examine to MATLAB because of the very fact the raspberry pi run at 900 MHz frequency therefore the rate of convergence is excessive and to boot the speed an excessive amount of low. The hassle with this machine is that the affected person needs to choose mode manually, there could also be no processed classification of alerts. therefore in destiny, style the artificial neural community within the python for the automated classification of the speech signal while patient go from one scheme to the opposite scheme.

VIII. FUTURE SCOPE

In the Future scope of this venture, as critical DSP processor C2000 processor is also wont to growth the general performance of the machine. C2000 devices are 32-bit microcontrollers with excessive overall performance enclosed peripherals designed for period management programs. Its optimized centre will run over one complicated management algorithms at speeds essential for worrying manipulate programs. effective enclosed peripherals intermingled with the SPI, UART (SCI), I2C, CAN, and McBSP communication peripherals build C2000 devices the acceptable single-chip manipulate answer. In destiny, it's far viable to layout the artificial neural community within the python for the automatic class of the speech signal whilst affected person go from one ecosystem to the alternative environment. similarly this setup will be also related to internet as raspberry pi acts as mini laptop.

REFERENCES

[1] Shankarayya G. Kambalimath Dept. of Electronics & Comm. Engg.,, Prem C. PandeyDept. of Electrical Engg., Pandurangarao N. KulkarniShivaling S. Mahant-ShettiDept. of Electronics & Comm. Engg "FPGA Based Implementation of Comb Filters for Use in Binaural Hearing Aids for Reducing Intraspeech Spectral Masking"2013 Annual IEEE India Conference (INDICON)P. N. Kulkarni1, P. C. Pandey2, and D. S. Jangamashetti,"Multi-Band Frequency Compressionfor Sensorineural Hearing Impairment"2009 IEEE

[2] Chaudhari D.S., Pandey P.C., 2001 IEEE. "Dichotic presentation of speech signal with critical band filtering for improving speech perception". In: Proc. ICASSP. Seattle, Washington, pp. 3601–3604.

[3] Deshmukh, S. G., Wani, M. G. Patil, S. A. "Improving Speech Perception of Sensorineural Hearing Impaired Using DSP TMS320C6713".Int. J. Computer & Electronics Engg.(Vol.IV,pp.224-228)

[4] Koichi Yanagisawa", Kyoko Tanaka, and ItsuoYamaura,"Graduate School of Science and Technology, Shinshu University, 3-15-1 Tokida, Ueda 386-8567,JAPA,Faculty of Textile Science and Technology, Shinshu University,3-15-1 Tokida, Ueda 386-8567, JAPAN,"Applying Comb Filter to Noise Reduction of Hearing Aid"1999 IEEE

[5] Alice N. Cheeran' and Prem C. Pandey' 'BME Group, IIT Bombay, Powai Mumbai-400 076. India .'EEE Dept., IIT Bombay, Powai Mumbai-400 076, India," Optimizing The Sweep Cycle Of Time-Varying Comb Filters For Binaural Dichotic Presentationin Sensorineural Hearing Impairment".2002 IEEE

[6] Kunal P. Ambhore, Dr. Mahesh T. Kolte," FPGA Based Signal Processing Implementation for Hearing Impairment" International Journal of Emerging Engineering Research and Technology Volume 2, Issue 2, May 2014, PP 160-164 S.G. Deshmukh1, S.A. Pangaonkar2 and M.K. Datir "Effective Filter Design Using FDA Tool for Improving Speech Intelligibility of Hearing Impaired International Journal of Information and Computation Technology. ISSN 0974-2239 Volume 4, Number 4 (2014), pp. 419-424

[7] Alice N. Cheeran, Prem C. Pandey, and Dakshayani S. Jangamashetti,"DESIGN OF COMB FILTERS BASED ON AUDITORY FILTER BANDWIDTHS FOR BINAURAL DICHOTIC PRESENTATION FOR PERSONS WITH SENSORINEURAL HEARING IMPAIRMENT'BME" Group, IIT Bombay, Powai Mumbai-400 076, India,2002 IEEE

[8] D. S. Chaudhari1 and P. C. Pandey School of Biomedical Engineering Department of Electrical Engineering Indian Institute of Technology, Bombay Powai, Mumbai 400 076, India," Critical Band Splitting Of Speech Signal For Reducing The Effect Of Spectral Masking In Bilateral Sensorineural Hearing Impairment, 5th International Symposium on Signal Processing and its Applications Brisbane, Australia, 23rd - 25th August, 1999

[9] Cheeran, A. N., &Pandey, P. C. (2004). "Speech processing for hearing aids for moderate bilateral sensorineural hearing loss". In Proceedings of the IEEE international conference on acoustics, speech and signal processing(Vol. IV,pp. 17–20).

[10] David Halupka, Member, IEEE, Alireza Seyed Rabi, Parham Aarabi, Senior Member, IEEE, and Ali Sheikholeslami, Senior Member, IEEE,"Low-Power Dual-Microphone Speech Enhancement Using Field Programmable Gate Arrays" IEEE TRANSACTIONS ON SIGNAL PROCESSING, VOL. 55, NO. 7, JULY 2007

[11] P. N. Kulkarni1, P. C. Pandey2, and D. S. Jangamashetti,"Study Of Perceptual Balance In Comb Filter Based Spectral Splitting Of Speech Signal To Reduce The Effect Of Frequency Masking"Int. Symposium on Frontiers of Research on Speech and Music 2008 (FRSM 2008), Kolkata, India, pp. 101-105.

[12] N. Kehtarnavaz and S. Mahotra, "FPGA implementation made easy for applied digital signal processing courses," in Proc. IEEE ICASSP, 2011, pp. 2892 – 2895.

[13] Cenk Demiroglu, David V. Anderson, Mark. A. Clements, and Thomas BarnwellDepartment of Electrical and Computer EngineeringGeorgia Institute of Technology, USA," Multi-Sensor Spectro-Temporal Comb Filtering for Speech Enhancement" 2007 IEEE

[14] N. S. Lawand, W. Ngamkham, G. Nazarian, P. J. French, W. A. Serdijn, G. N. Gaydadjiev,J. J. Briaire, and J. H. M. Frijns." An improved system approach towards future cochlear implants." 35th Annual International Conference of the IEEE EMBS Osaka, Japan, 3 - 7 July, 2013

[15] Bo Li, Student Member, IEEE, and Khe Chai Sim, Member, IEEE,"A Spectral Masking Approach to Noise-Robust Speech Recognition Using Deep Neural Networks" IEEE/ACM Transactions On Audio, Speech, And Language Processing, Vol. 22, NO. 8, AUGUST 2014

[16] S. G. Kambalimath, P. C. Pandey, P. N. Kulkarni, S. S. Mahant-Shetti, and S. G. Hiremath, "FPGA based implementation of comb filters for Using Sequential Multiply-Accumulate Operations for Use in Binaural Hearing Aids," in Proc. IEEE INDICON, 2014 Annual IEEE India Conference (INDICON)

[17] A.V.Vikhe,U.V.Patil," Real Time Noise Cancellation and Improving Speech Intelligibility of Sensorineural Hearing Impaired by Using DSP Processor"Electronics Engineering Post Graduate Students Project Exhibition,June 2016.

[18] A.V.Vikhe,U.V.Patil," Improving Speech Intelligibility of Sensorineural Hearing Impaired and Real Time Noise Cancellation Using Filters ", International Journal of Informative and Futuristic Research. ISSN 2347-1697 Volume 3, Issue 10,June 2016, pp.3856-3866 Sources of Information: www.google.com www.wikipedia.com, www.hearingloss.org