4th International Conference on Recent Innovations in Science Engineering and Management
India International Centre, New Delhi
(ICRISEM-16)
20th March 2016, www.conferenceworld.in
ISBN: 978-81-932074-6-8

EXISTING NOISE REMOVAL TECHNIQUES USED IN COCHLEAR IMPLANT

Deepti Gupta[1], Dr. Pratistha Mathur[2], Dr. Peeyush Tewari[3]

[1]Research Scholar, Banasthali Vidyapith University Rajasthan (India)

[2]Associate Professor, Department of Computer Science,
Banasthali vidyapith University, Rajasthan (India)

[3]Associate Professor, Department of Mathematics Birla Institute of Technology Noida, U.P. (India)

ABSTRACT

Cochlear Implant is a technology used to give hearing power to profoundly deaf people. Cochlear Implant process also helps to give solution in different region of disability in hearing. Existing Cochlear Implant techniques can't satisfy all type of deficiency in hearing but still gave a big relief to people. This paper is a review of how technology has changed in the past decade and how can find better techniques for noise removal consistently. Initially Subspace algorithm with single channel was proposed which works on stationary noise removal and it changed the lives of the people and children by developing the speech and language skills prior to deterioration of their hearing. Initially people faced lots of problem with regard to noise that interfered with the speech signal, so single microphone noise reduction algorithms came into existence because mostly work was done with speech quality. Further focus was based on sound coding strategies for suppressing the noise in CI. Likewise various technologies came into existence to enhance the quality of CI and for better results given to the users. This survey addressed various coherent noise removal improvement techniques and their evolution and also presents the issues related to those techniques.

Keywords: Cochlear Implant (CI), Time-Frequency (T-F), Bilateral Cochlear Implants (BI-CIs), Dual-microphone (DM).

I. INTRODUCTION

There are various reasons for hearing loss of humans which are noise, aging, disease, and heredity. Hearing loss is of three types conductive and sensorineural and mixed. Countries like France, Austria, Australia and the United States, etc. have worked on Cochlear implants [1, 2] and made humans to hear normally with after suitable implant and rehabilitation through speech training. Area such as bioengineering, physiology, otolaryngology, speech science, signal processing discipline contributed to the design of various aspects for Cochlear Implant. The Cochlear implant designers faced a lot of problems in developing signal processing technique [11, 12] that would mimic the function of normal cochlea of inner ear. There is a need to understand the working of auditory system before the development of Cochlear implants because initially needed to know how a normal auditory system functions [9, 10, 3, 4].

However, it is not possible to implant this in each and every case. On the basis of some criteria one could use Cochlear implant CI which are when hair cells of inner ear or some auditory nerve get damaged then only can replace it by CI. But if total nerves get damaged then there is no use of CI because Cochlear implant enables the sound to be transferred to our hearing nerves and enable us to hear. Cochlear implant developers aimed at stimulation of remaining neurons which could be excited directly through electrical. Because of this reason Cochlear implant is developed by bypassing the normal hearing mechanism and electrically stimulating the auditory neurons [13, 14, 15]. But the main challenge is to finding out how to stimulate auditory neurons for meaningful information about speech which is conveyed to the brain [6].

II. BACKGROUND

Sound travels through the outer ear, middle ear, inner ear, auditory nerve and into the brain as a series of transformations [16, 17], [Van Hoesel and Clark (1995) Electrical Stimulation, 18]. The outer ear picks up acoustic pressure waves that are converted to mechanical vibrations by a series of small bones in the middle ear. In the inner ear, the cochlea, a snail-shaped cavity filled with fluid, transforms the mechanical vibrations to vibrations in fluid. Pressure variations within the fluids of the cochlea lead to displacements of a flexible membrane, called the basilar membrane, in the cochlea. These displacements contain information about the frequency of the acoustic signal. Attached to the basilar membrane are hair cells that are bent according to the displacements of the basilar membrane. The bending of the hairs releases an electrochemical substance that causes neurons to fire, signaling the presence of excitation at a particular site in the inner ear. These neurons communicate with the central nervous system and transmit information about the acoustic signal to the brain [5]. In Cochlear implant, a sound processor placed behind the ear which captures the sound and turns it into digital code. The sound processor has a battery that by which can take powers for the entire system. The sound processor transmits the digitally-coded sound through the coil on the outside of your head to the implant. The implant converts the digitally-coded sound into electrical impulses and sends them along the electrode array placed in the cochlea (the inner ear). The implant's electrodes stimulate the cochlea's hearing nerve, which then sends the impulses to the brain where they are interpreted as sound. We can understand this phenomenon by the given figure-1 of the Cochlear implant [7, 8].

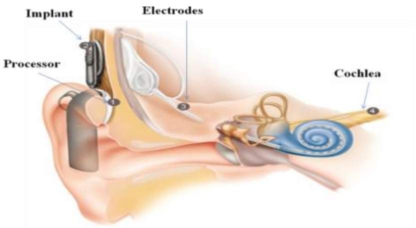

Fig-1 Cochlear Implant Placement in Ear

III. COCHLEAR IMPLANT TECHNIQUES

Many Cochlear implant techniques like subspace algorithm, single microphone algorithm, etc been developed by researchers and scientists, some of the most important and latest research work on Cochlear implant techniques highlighted with the help of discussion and evaluation.

3.1 Subspace Algorithms

Subspace algorithms function at single-channel algorithms which provide sentence recognition in stationary noise. The assumption in this algorithm was based on the two or more microphones were available. So two microphones give benefit to the CI user in moderate reverberation condition. The algorithm is based on the projection of the noisy speech vector onto two subspaces: the "signal" subspace and the "noise" subspace.

Later on further work on this algorithm did by Van Hoesel and Clark [19] also in 1995. After that approach is proposed for enhancement of speech corrupted by coloured noise in 2003 by Yi Hu [20] and further based on the subtraction of the noisy speech envelopes from an estimate of the noise envelopes in 2003 by F. Toledo, P. Loizou and A. Lobo [21]. A multichannel noise reduction methods are used by Yiteng Huang, Jacob Benesty, Jingdong Chen in 2008 it introduces speech distortion to the desired speech signal while reducing noise. This is a combination of three algorithms in which subspace algorithm working also included [22]. Basically CI is not able to work in real time noisy environment in that respect in 2012 another algorithms was implemented which is capable of classifying the background noise environment in real time [23]. But it is not suitable for non-stationary environments. So, further changes are needed to extend the subspace algorithm to non-stationary noise environments [58, 59].

3.2 Single Microphone noise reduction algorithm

There are four algorithms present spectral subtractive, subspace, statistical-model based and wiener-type. The voice activity detector (VAD) which is based on statistical-model is used in all algorithms to update the noise spectrum during non-speech periods. Speech intelligibility is evaluated in terms of identification for correct words percentage in noise. The words which are correctly recognized give the counting for scoring.

The algorithms which have been evaluated for intelligibility tests by Boll in 1979 [24] by Lim in 1978 [25]; by Tsoukalas in 1997 [26];by Arehart in 2003 [27]. Noise cancellation algorithms (passive or active) work for decreasing unwanted ambient sound [28, 29] and Ideal binary mask decomposes the input signal into time-frequency (T-F) units and makes binary decisions means whether each T-F unit is dominated by the target or the masker. It gives substantial improvements in intelligibility [30]. Discrete Fourier transform (DFT) domain is used to improve on existing single-microphone schemes for an extended range of noise types and noise levels [31]. From the comparison of the performance of algorithms, it found that the Wiener-as algorithm performed is good in all conditions for both sentence and consonant recognition tasks. On consonant recognition, the Wiener-as, KLT, and WavThr algorithms performed equally well [60, 61].

3.3 Sound Coding Strategy for Noise Suppression

Advanced combination encoder (ACE), and Continuous interleaved sampling (CIS) strategies are speech coding techniques which are based on channel vocoder principles, in which by the signal extract the envelopes from each band. Selection of number of envelopes for stimulation differs in the CIS and ACE strategies. In ACE strategy, only a subset n out of 22 envelopes is selected and used for stimulation purpose at very cycle and all 22 electrode sites are utilized for stimulation but in CIS strategy, only a fixed number of envelopes are computed, and only the corresponding electrode sites are used for stimulation purpose.

There are various studies which shows a high gains in intelligibility in noise with the help of ACE coding as before by some researchers namely McDermott in 1992 [32]; McKay and McDermott in 1993 [33]; Vandali in 2000 [34]; Kiefer in 2001[35]; Loizou in 2006 [36]. Although ACE works well in quite listening conditions but its quality decreases by reverberation but we also did work in that platform and now we can use ACE without reverberation [37]. But there are some deficiency in ACE strategy, while it offers the long duration battery life because all electrodes need not to be stimulated at a given instant but in the case of noise, this criterion could be problematic. Thus there is a need of better selection criterion to compensate for the improvement of ACE in noise. Thus an algorithm which is capable of estimating the accurate SNR (sound noise ratio), can give significant gains in intelligibility [62, 63].

3.4 Bilateral Cochlear implant in party

Bilateral Cochlear implantation gives advantage of listening with two ears. Speech intelligibility gets reduced when noise sources are placed symmetrically and increased when they are placed asymmetrically [38, 39]. Based on masker effect there is a need to evaluate performance with both speech and non-speech maskers of bilateral CI (BI-CI) users. In the binaural hearing we use both ears but the binaural-interaction benefit is quite small, often not significant, and variable according to users. The lack of binaural advantage in bilateral CIs affects several factors which are poor ITD sensitivity, poor spectral resolution and asymmetry in the state of the binaural auditory pathways. Bilateral CI users may not get benefits from binaural hearing may be due to the etiology of hearing loss because it might differ in the two ears. The electrode insertion depth in the two ears might be differing. Some people can argue that such a mismatch in insertion depth might be beneficial, but can be quite harmful to the mechanisms involved in processing ITD information.

The non-energetic masking, usually called informational masking, is produce confusion because of content similarity between the target and the interferer. The informational masking is reduced when the target and interferer signals are spatially separated. But it was not found in the case of with bilateral CI users. The benefit is roughly same in both the cases from spatial separation either noise or speech interferers [64, 65], [40, 41].

3.5 Multi-microphone in Bilateral Cochlear implant

In last few years single microphone noise reduction techniques proposed for noisy background conditions and we found better speech intelligibility results but found better improvement when used multi-microphone instead of single microphone and better exploit the condition in which the target and masker are spatially separated.

From the above studies evaluated beam forming algorithms in situations where only a single interfering source was present and the room acoustics were characterized with low or no resonance. While in real condition rooms can have moderate to high resonance and multiple noise sources can be present. So now talk about on the multi-microphone algorithms based on that can focus on two strategies of signal processing.

Signal processing strategies i.e. 2M2-BEAM strategy is a noise reduction strategy which utilize the dual-microphone concept BEAM running in an independent stimulation mode in both ears. Comparing the 2M2-BEAM processing strategy against the baseline bilateral unprocessed condition, it found there is a marginal improvement in speech intelligibility, while the 4M-SESS strategy yielded a considerable benefit in all conditions [66, 67]. It is possible that with the help of research processors it can carefully control the stimulus delivered to each electrode in each ear, so that able to preserve binaural cues and can deliver them to BI-CI users [42, 43].

3.6 Environment specific noise suppression

Various noise reduction algorithms had been defined for unilateral CI users. In the spectral channel noisy envelopes were present. These noise suppression methods gave lots of improvements in intelligibility. Instead of coding strategies now focus on the environmental-specific noise suppression algorithms which can be implemented in two ways.

One way is programming the speech processors with multiple MAPs where every program designed for different situation. Second way is to include a sound classification algorithm, which automatically identify the listening environment and so switch the program accordingly. The study can evaluates the performance of CI users by a noise suppression algorithm, defined for three different real-world environments, namely exhibition hall, multi-talker babble and train. Two main reasons gave the high performance of the GMM-based noise reduction algorithm: first, one AMS-like features are neuro-physiologically. 2nd one GMM-based Bayesian classifiers are beneficial for binary mask application. Neural networks can be used as an alternative.

There are several points of discussion which require further improvement of the GMM-based noise reduction approach. First is the speaker gender had no significant impact on performance, while require this for extracting features and for carry much information for the identification purpose of the speaker. Secondly, it used an FFT-based feature extraction process and the envelope segment i.e. 20Hz was not enough to maintain modulations because below 20 Hz is necessary for speech intelligibility. A better solution is to use a wavelet-based feature extraction procedure which is based on the use of different window lengths of different frequency components. 3rd GMM-based noise reduction method needs further improvement [68, 69]. There are various noise reduction algorithms for unilateral CI users have been proposed by Hochberg in 1992 [44]; Weiss, in 1993 [45]; Yang and Fu, in 2005 [46]; Loizou in., 2005 [47]; Kasturi and Loizou,in 2007[48]; Hu in 2007[49] and Yang and Fu in 2005 [50], and some models also which are Gaussian mixture models (GMM), support vector machines (SVM), and neural networks (NN)[51].

3.7 Dual microphone in Bilateral Cochlear implant

Many noise reduction algorithms are implemented. Here discuss about only three types of noise for enhancement in speech. 1st one is incoherent noise, 2nd is coherent noise and 3rd is diffuse noise.

Earlier only one microphone is used but now using array based microphone. By using multiple microphone further think about reducing the noise but its size, weight and power consumption is a big difficulty so used dual-microphone (DM) as a speech enhancement system. Beamforming is the most common algorithms in this field. Two common fixed beamformers are the delay-and-sum and super-directive.

In 2005, the beamformer was implemented behind the ear speech processor used in CI system. It worked better when only single noise source present but performance decreases in the presence of multiple noise sources. Initially suppose that the noise and target speech signals are spatially separated. Thus for improvement purpose used a dual-microphone speech enhancement technique that is based on the magnitude of coherence between input signals.

Dual microphone (DM) algorithm is computed on the real and imaginary parts of the coherence function. In this algorithms there is no assumption regarding the placement of the noise sources. The result of the proposed algorithm showed the higher intelligibility and quality than that obtained by the beam former, especially in multiple noise-source scenarios and competing talkers [70, 71]. To decrease background noise and speech distortion and increase speech quality, researchers utilized multichannel microphones to exploit all available acoustic and spatial information of the speech and noise sources [52].

3.8 Channel selection modulation for intelligibility

The speech signal can be defined as a sum of amplitude-modulated signals. The output waveforms of each sub-band can be defined in terms of a carrier signal and an envelope. The temporal modulations present in the envelope gives important information of both segmental means, manner of articulation and suprasegmental means intonation for distinctions in speech.

Modulation frequencies between 4 and 16 Hz were achieved the most to intelligibility, with the region around 4–5 Hz being the most significant, and reflects the rate at which syllables are produced. Now consider the selection of target-relevant modulations from the corrupted target and masker envelopes also for designing algorithms that potentially improve speech intelligibility. As a selection criterion consideration of the signal-to-noise ratio defined in the modulation spectral domain, denoted by $(S/N)_{mod}$ for distinguishing purpose from the signal-to-noise ratio (SNR) as defined in acoustic spectral domain. On this modulation-selective criterion, envelopes can be made by retaining modulations with $(S/N)_{mod}$ greater than a defined threshold, and discarding modulations with $(S/N)_{mod}$ less than a defined threshold.

In this work, suppose a priori knowledge of $(S/N)_{mod}$ present for prior to mixing of the target and masker. The modulation spectra allow processing in the modulation domain on relatively short intervals of 256ms. This is done for common practices of extremely long means in the form of minute's speech segments from continuous discourse to calculate the modulation spectra.

The modulation channel-selection (MCS) method uses a dual analysis-modification-synthesis frame work that allows processing in the short-time modulation spectral domain [72, 73].

The inverse short-time Fourier transform is calculated for the modified modulation spectrum and the overlap-and-add procedure is used to give the modified trajectories of the acoustic magnitude spectrum. An inverse short-time Fourier transform of the acoustic spectrum is calculated and the overlap-and-add procedure is finally defined to synthesize the speech signal. Another approach for channel selection is used for selecting reliable channels in which selection criterion is based on operating in the short-term modulation spectrum domain. This method quantifies the relative strength of speech from each microphone and speech obtained from beamforming modulations [53].

3.9 Time- Frequency contribution in speech intelligibility

An ideal time-frequency (T-F) binary mask produces computational auditory scene analysis (CASA). In 1st method speech sentences were produced by speaker in a soundproof booth and then sample it after that down sampled it. A multi-talker babble noise source was used as the masker to corrupt the sentences.

The T-F representations were composed with T-F units having equal area, and the length along time and the width along frequencies. The T-F analyzed signals were pre-emphasized by an equal-loudness curve. The correlation between intensity and perceptual loudness of sound was then modeled by using a power law compression.

In order to detect speech activity in a given T-F unit, the energy of the required signal of the given T-F unit was compared with floor value. For each sentence, this floor level was selected separately within each frequency band to get 95% target loudness of individual frequency band. Speech-present T-F units were defined by a value of 1, while speech-absent T-F units were defined by a value of 0.

In 2nd method new binary masks were created by introducing a fixed percentage of miss errors and false alarm errors. For identification purpose of the effect of miss errors, a fixed percentage of speech-present T-F units defined as 1 in experiment were flipped to 0, while no mask errors were created on speech-absent T-F units. In the same way, for identification purpose of the effect of false alarm errors, a fixed percentage of speech-absent T-F units labeled as 0 in experiment were flipped to 1, while no mask errors were created on speech-present T-F units. This type of Stimuli was created from the new binary masks containing the fixed rate of miss or false alarm errors [76, 77].

In the new method which further works on that algorithm examine the effects of three noises on supervised speech separation: noise rate, vocal tract length, and frequency uneasiness at low signal-to-noise ratios (SNRs) [54].

The Phase Error Filtering (PEF) algorithm managing a Time-Frequency (TF) mask for noise reduction [55].

3.10 Bilateral Cochlear implant with single processor

Bilateral Cochlear implants increase the reliability of the system with the advantage of the two signal sources in the form of two ears, and provide better enhancement of the noisy signals. While in unilateral CIs, there is no directional information received by patients, so they face difficulties in localization of sound sources.

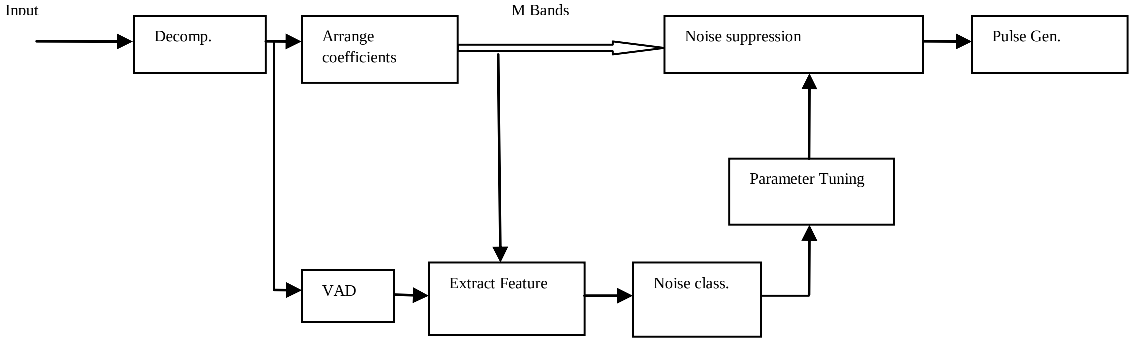

Fig-2 Block diagram of the unilateral Cochlear implant system

Figure 2 defines the environment-adaptive pipeline for unilateral Cochlear implant. A Voice Activity Detector (VAD) is used to determine signal frames containing speech. When it is purely noise, a Gaussian Mixture Model (GMM), trained on different noise classes, is used to determine the noise type.

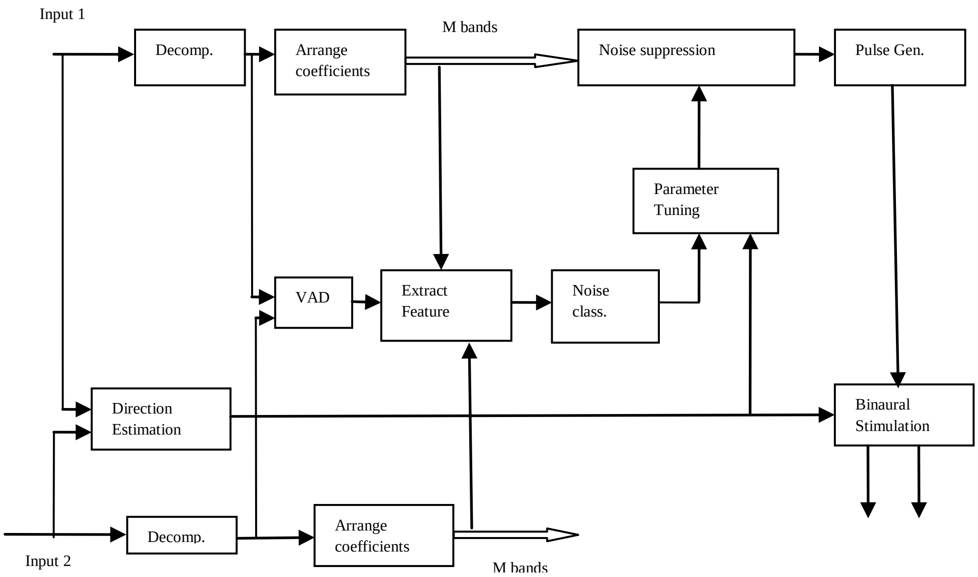

Fig-3 Block diagram of bilateral Cochlear implant system

In figure 3 the parameter space consisting of noise suppression parameters and directional parameters also. And gains are considered to be functions of both direction and SNRs. The identification of gain function becomes different along different directions.

The reference signal is the first input if the delay $\hat{\tau}$ is positive, and it is the second one when it is negative.

For comparison purpose of the bilateral hearing conditions, speech sentences were convolved with a patient-specific Head-Related Transfer Function (HRTF). Training of the suppression functions along with the HRTF gain parameters were performed and HRTF gain parameters associated with each environment were trained by adding the recorded noise samples to the given sentences as clean speech signals. The resulting noisy files were then used to generate the required training set [74, 75].

A different framework is introduced that allows one to train suppression and head-related transfer function gain not only for different noise environments but also for different distortion measures [56].The use of only a single processor to provide binaural stimulation signals overcomes the synchronization problem, which is an existing challenging problem in the deployment of bilateral CI devices [57].

IV. CONCLUSION

The main objective of above discussed algorithms was to enhance speech quality usually limited no. of noise conditions is modeled and no statistical test runs on data are performed. All these reasons account for not getting the optimum performance. There are various types of noise that can be identified and needed to be removed from the speech identification, which makes it as distant goal of achieving the perfect speech quality being given to the user so that user feels the real sound as normal hearing person feels.

REFERENCES

[1]. https://en.wikipedia.org/wiki/Cochlear_implant.

[2]. http://www.Cochlear.com/wps/wcm/connect/au/about/Cochlear-30-anniversary

[3]. J. Millar, Y. Tong, and G. Clark, "Speech processing for Cochlear implant prostheses,"Journal of Speech and Hearing Research, vol. 27, pp. 280-296,1984.

[4]. C. Parkins and S. Anderson (eds.), Cochlear prostheses: An international symposium.New York: New York Academy of Sciences, 1983.

[5]. W. Yost and D. Nielsen, Fundamentals of Hearing: An Introduction. New York: Holt, Rinehart and Winston, 1985.

[6]. R. Hinojosa and M. Marion, "Histopathology of profound sensorineural deafness," Annals of New York Academy of Sciences, vol. 405, pp. 459-484,

[7]. https://en.wikipedia.org/wiki/Cochlear_implant

[8]. http://www.Cochlear.com/wps/wcm/connect/au/home/understand/hearing-and-hl/hl-treatments/Cochlear-implant

[9]. B. Wilson, "Signal processing," in Cochlear Implants: Audiological Foundations (R. Tyler, ed.), pp. 35-86, Singular Publishing Group, Inc, 1993.

[10]. G. Loeb, "Cochlear prosthetics," Annual Review in Neuroscience, vol. 13, pp.357-371, 1990.

[11]. G. Fant, Acoustic Theory of Speech Production. The Hague, Netherlands: Mouton, 1970.

[12]. G. Borden, K. Harris, and L. Raphael, Speech Science Primer: Physiology, Acoustics, and Perception ofSpeech. Baltimore, Maryland: Williams and Wilkins, 1994.

[13]. http://www.sciencedirect.com/science/article/pii/0378595583901156.

[14]. http://www.ncbi.nlm.nih.gov/pubmed/6619003.

[15]. http://www.ncbi.nlm.nih.gov/pubmed/6689326

[16]. http://www.ncbi.nlm.nih.gov/books/NBK20366/

[17]. http://www.explainthatstuff.com/sound.html

[18]. http://www.ncbi.nlm.nih.gov/pmc/articles/PMC1343472/#R1

[19]. http://www.ncbi.nlm.nih.gov/pmc/articles/PMC1343472/#R4

[20]. http://ieeexplore.ieee.org/stamp/stamp.jsp?tp=&arnumber=1214849

[21]. http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.125.5074&rep=rep1&type=pdf

[22]. http://externe.emt.inrs.ca/users/benesty/papers/aslp_jul2008.pdf

[23]. http://ieeexplore.ieee.org/stamp/stamp.jsp?tp=&arnumber=6175112

[24]. Boll, S. F. _1979_. “Suppression of acoustic noise in speech using spectral subtraction,” IEEE Trans. Acoust., Speech, Signal Process. ASSP-27,113–120.

[25]. Lim, J. S. _1978_. “Evaluation of a correlation subtraction method for enhancingspeech degraded by additive white noise,” IEEE Trans. Acoust.,Speech, Signal Process. 26, 471–472.

[26]. Tsoukalas, D. E., Mourjopoulos, J. N., and Kokkinakis, G. _1997_. “Speech enhancement based on audible noise suppression,” IEEE Trans. Speech Audio Process. 5, 479–514.

[27]. Arehart, K. H., Hansen, J. H., Gallant, S., and Kalstein, L. _2003_. “Evaluation of an auditory masked threshold noise suppression algorithm in normal-hearing and hearing-impaired listeners,” Speech Commun. 40,575–592.

[28]. Carnoy, David, Steve Guttenberg, and Eliot Van Buskirk. "The Sound of Silence." CNET Reviews. 19 July 2005. 12 July 2007 <http://reviews.cnet.com/4520-3000_7-1017728-1.html>.

[29]. http://www4.ncsu.edu/~rsmith/MA574_S15/silence.pdf

[30]. http://ecs.utdallas.edu/loizou/cimplants/GMM_intelligib_sept09.pdf

[31]. http://mmc.tudelft.nl/sites/default/files/phdthesis_5_1_08.pdf

[32]. McDermott, H. J., McKay, C. M., and Vandali, A. E. (1992). “A new portable sound processor for the University of Melbourne=Nucleus Limited multichannel Cochlear implant,” J. Acoust. Soc. Am. 91, 3367–3371.

[33]. McKay, C. M., and McDermott, H. J. (1993). “Perceptual performance of subjects with Cochlear implants using the Spectral Maxima Sound Processor (SMSP) and the Mini Speech Processor (MSP),” Ear Hear. 14, 350–367.

[34]. Vandali, A. E., Whitford, L. A., Plant, K. L., and Clark, G. M. (2000). “Speech perception as a function of electrical stimulation rate: Using the Nucleus 24 Cochlear implant system,” Ear Hear. 21, 608–624.

[35]. Kiefer, J., Hohl, S., Sturzebecher, E., Pfennigdorff, T., and Gstoettner, W.(2001). “Comparison of speech recognition with different speech coding strategies (SPEAK, CIS, and ACE) and their relationship to telemetric measures of compound action potentials in the Nucleus CI 24M Cochlear implant system,” Audiology 40, 32–42.

[36]. Loizou, P. C. (2006). “Speech processing in vocoder-centric Cochlear implants,” in Cochlear and Brainstem Implants, edited by A. Moller (Karger, Basel, Switzerland), Vol. 64, pp. 109–143.

[37]. http://ecs.utdallas.edu/loizou/cimplants/reverb_chann_selection_jasa2011.pdf

[38]. Bronkhorst A. W., and Plomp R. (1992). “Effect of multiple speechlike maskers on binaural speech recognition in normal and impaired hearing,” J. Acoust. Soc. Am. 92, 3132–3139.10.1121/1.404209

[39]. Yost W., Dye R., and Sheft S. (1996). “A simulated “cocktail party” with up to three sound sources,”Percept. Psychophys. 58, 1026–1036.

[40]. nov_2011http://digitalcommons.wustl.edu/cgi/viewcontent.cgi?article=1007&context=audio_hapubs

[41]. file:///C:/Documents%20and%20Settings/CRL702A/My%20Documents/Downloads/IJCM_2013032814374000.pdf

[42]. http://www.plosone.org/article/fetchObject.action?uri=info:doi/10.1371/journal.pone.0126133&representation=PDF

[43]. http://www.waisman.wisc.edu/bhl/about_publications/2012LitovskyJAAA.pdf

[44]. Hochberg I., Boorthroyd A., Weiss M., and Hellman S. (1992). “Effects of noise and noise suppression on speech perception by Cochlear implant users,” Ear Hear. 13, 263–271.10.1097/00003446-199208000-00008

[45]. Weiss M. (1993). “Effects of noise and noise reduction processing on the operation of the nucleus-22 Cochlear implant processor,” J. Rehabil. Res. Dev. 30, 117–128.

[46]. Yang L., and Fu Q. (2005). "Spectral subtraction-based speech enhancement for Cochlear implant patients in background noise," J. Acoust. Soc. Am. 117, 1001–1003.10.1121/1.1852873

[47]. Loizou P., Lobo A., and Hu Y. (2005). "Subspace algorithms for noise reduction in Cochlear implants," J. Acoust. Soc. Am. 118, 2791–2793.10.1121/1.2065847

[48]. Kasturi K., and Loizou P. (2007). "Use of s-shaped input-output functions for noise suppression in Cochlear implants," Ear Hear. 28, 402–411.10.1097/AUD.0b013e31804793c4

[49]. Hu Y., Loizou P., Li N., and Kasturi K. (2007). "Use of a sigmoidal-shaped function for noise attenuation in Cochlear implant," J. Acoust. Soc. Am. 122, EL128–EL134.10.1121/1.2772401

[50]. Yang L., and Fu Q. (2005). "Spectral subtraction-based speech enhancement for Cochlear implant patients in background noise," J. Acoust. Soc. Am. 117,10011003.10.1121/1.1852873.

[51]. http://www.mirlab.org/conference_papers/International_Conference/ICASSP%202014/papers/p900-hazrati.pdf

[52]. J. Polastre, R. Szewczyk, A. Mainwaring, Chapter 18: Analysis of Wireless Sensor Networks for Habitat Monitoring, in Wireless Sensor Networks, Kluwer Academic Publishers, Norwell, MA, 2004

[53]. http://infoscience.epfl.ch/record/209075/files/Himawan_INTERSPEECH_2015.pdf

[54]. ftp://ftp.cse.ohio-state.edu/pub/tech-report/2014/TR16.pdf

[55]. ftp://ftp.esat.kuleuven.be/pub/SISTA/ida/reports/12-21.pdf

[56]. http://ieeexplore.ieee.org/stamp/stamp.jsp?tp=&arnumber=6639074&tag=1

[57]. http://arxiv.org/ftp/arxiv/papers/1409/1409.6554.pdf

[58]. 1983.Ephraim Y, Van Trees HL. "A signal subspace approach for speech enhancement," IEEE Trans Speech Audio Proc. 1995;3:251–266.

[59]. Hamacher V, Doering W, Mauer G, Fleischmann H, Hennecke J. "Evaluation of noise reduction systems for Cochlear implant users in different acoustic environments," Am J Otol.1997;18:S46–S49.

[60]. Arehart, K. H., Hansen, J. H., Gallant, S., and Kalstein, L. _2003_. "Evaluation of an auditory masked threshold noise suppression algorithm in normal-hearing and hearing-impaired listeners," Speech Commun. 40,575–592.

[61]. Boll, S. F. _1979_. "Suppression of acoustic noise in speech using spectral subtraction," IEEE Trans. Acoust., Speech, Signal Process. ASSP-27,113–120.

[62]. ANSI (1997). "Methods for calculation of the speech intelligibility index," ANSI S3.5-1997, American National Standards Institute, New York.

[63]. 16.Anzalone M., Calandruccio L., Doherty K., and Carney L. (2006). "Determination of the potential benefit of time-frequency gain manipulation," Ear Hear. 27, 480–492.10.1097/01.aud.0000233891.86809.df

[64]. Blauert J., Brueggen M., Bronkhorst A. W., Drullman R., Reynaud G., Pellieux L., Krebber W., and Sottek R. (1998). "The AUDIS catalog of human HRTFs," J. Acoust. Soc. Am. 103, 3082.

[65]. Bronkhorst A. (2000). "The cocktail party phenomenon: A review of research on speech intelligibility in multiple-talker conditions," Acta Acust. 86, 117–128.

[66]. Chung K., Zeng F. -G., and Acker K. N. (2006). “Effects of directional microphone and adaptive multichannel noise reduction algorithm on Cochlear implant performance,” J. Acoust. Soc. Am. 120, 2216–2227.10.1121/1.2258500

[67]. Ericson H., Svard I., Devert G., and Ekstrom L. (1988). “Contralateral routing of signals in unilateral hearing impairment: A better method of fitting,” Scand. Audiol. 17, 111–116.

[68]. Drullman R., Festen J., and Plomp R. (1994a). “Effect of reducing slow temporal modulations on speech reception,” J. Acoust. Soc. Am. 95, 2670–2680.10.1121/1.409836

[69]. Drullman R., Festen J., and Plomp R. (1994b). “Effect of temporal smearing on speech reception,” J. Acoust. Soc. Am. 95, 1053–1064.10.1121/1.408467

[70]. Spriet A, Van Deun L, Eftaxiadis K, Laneau J, Moonen M, van Dijk B, Van Wieringen A, Wouters J. Speech understanding in background noise with the two-microphone adaptive beamformer BEAM in the nucleus freedom Cochlear implant system. Ear Hear. 2007 Feb.28(1):62–72.

[71]. Chen J, Phua K, Shue L, Sun H. Performance evaluation of adaptive dual microphone systems. Speech Commun. 2009 Dec.51(12):1180–1193.

[72]. Atlas L., and Shamma S. A. (2003). “Joint acoustic and modulation frequency,” EURASIP J. Appl. Signal Process. 2003, 668–675. 10.1155/S1110865703305013

[73]. Bacon S. P., and Grantham D. W. (1989). “Modulation masking: Effects of modulation frequency, depth, and phase,” J. Acoust. Soc. Am. 85, 2575–2580. 10.1121/1.397751

[74]. Algazi VR, Duda RO, Thompson DM, Avendano C. The CIPIC HRTF database. Paper presented at the IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics; 2001. pp. 99–102.

[75]. Chen J, Benesty J, Huang Y. Time delay estimation in room acoustic environments: An overview.EURASIP J Appl Signal Process. 2006;26:19.

[76]. Anzalone M., Calandruccio L., Doherty K., and Carney L. (2006). “ Determination of the potential benefit of time-frequency gain manipulation,” Ear Hear. 27, 480–492.10.1097/01.aud.0000233891.86809.df

[77]. Bregman A. (1990). Auditory Scene Analysis (MIT Press, Cambridge, MA:), pp. 1–570.