2. LITERATURE REVIEW

In this chapter, a brief review of literature on speech signal characteristics, function of human ear and cochlea, cochlear implant, discrete wavelet transform, adaptive signal processing algorithms like LMS, NLMS, RLS and comparison for different adaptive algorithms for various values and voice activity detection algorithm is covered.

- The technical article on ‘A Comparison of Noise Reduction Techniques for Speech Recognition in Telecommunications Environments’ shows how human speech carries different two types of sounds like voiced sound and unvoiced sound. The production and characteristics of the acoustic speech signal upon which recognition systems operate is well understood and can be modeled quite accurately. Paper also shows the physiology of the human vocal tract imposes articulatory constraints on the range of sounds which may be generated. The sounds may be classified according to their excitation [6].

- The research paper on ‘Improved Noise Power Spectrum Density Estimation for Binaural Hearing Aids Operating in a Diffuse Noise Field Environment’ suggests that how its possible for current noise to prepare reduction algorithms which are monaural and are therefore intended for only bilateral hearing aids. Recently, binaural in contrast to monaural noise reduction schemes have been proposed, targeting future high-end binaural hearing aids. Those new types of hearing aids would allow the sharing of information signals received from both left and right hearing aid microphones (via a wireless link) to generate an output for the left and right ear. A novel approach for noise power spectral density estimator for binaural hearing aids operating in a diffuse noise field environment, by taking advantage of the left and right reference signals that will be accessible, as opposed to the single reference signal currently available in bilateral hearing aids [7].

- The technical paper on ‘Speech Processing for Cochlear Implants with the Discrete Wavelet Transform: Feasibility Study and Performance Evaluation’ suggests how differently from the traditional filter-bank spectral analysis strategies are available and its possible to analyses the speech signal by means of the discrete wavelet transform (DWT). Preliminary tests were conducted in order to compare the WT and the filter-bank analysis methods. Additionally, the intelligibility of the speech processed with the proposed WT
strategy was tested on normal hearing people by means of the acoustic simulations and a comparison was made with respect to traditional CI algorithms [8].

- The technical article on ‘Fixed Filter Implementation Of Feedback Cancellation For In-The-Ear Hearing Aids’ shows how acoustic feedback oscillation in hearing aids is a problem which often prevents an adequate amount of gain from being realized for some hearing aid users. Paper show the technique consists of modeling the feedback path around the amplifier and duplicating it with an FIR (Finite Impulse Response) filter. The output of this filter is inverted and summed with the input signal at the input to the amplifier, canceling the signal from the feedback path around the amplifier. This technique is applied in its most optimized form and also in several variations of a simplified form [9].

- The research article on ‘A novel approach for single microphone active noise cancellation’ presents a novel approach for subband feedback Active Noise Control. This paper presents a time domain algorithm for single sensor subband feedback ANC using relatively short fixed FIR filters to do the subband processing. The adaptive coefficients in the system are updated using a weight constrained NLMS algorithm for feedback ANC. The proposed subband algorithm had a significant performance advantage over the traditional single band ANC algorithm in terms of the rate of convergence and the noise attenuation that could be obtained [10].

- The research paper on ‘On the convergence behavior of the LMS and the normalized LMS algorithms’ shows how it’s happened that for the designing of filter very less knowledge of input is available, normalized LMS algorithm is potentially faster converging algorithm compared to LMS algorithm. Paper also describes response of the NLMS can be speeded up by using time varying step size. By white noise arbitrary input here prior information of the signal can be also predicted [11].

- The research paper on ‘Adaptive Filtering with Averaging in Noise Cancellation for Voice and Speech Recognition’ describes how it’s possible to make the trend towards real-time, low-bit-rate speech coders dictates current research efforts in speech compression. A method being evaluated uses wavelets for speech analysis and synthesis. Distinguishing between voiced and unvoiced speech, determining pitch, and methods for choosing optimum wavelets for speech compression are discussed. Paper also describes
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the optimum wavelets are selected based on energy conservation properties in the approximation part of the wavelet coefficients. The Haar wavelet is the worst. Listening tests show that the Daubechies 10 preserves perceptual information better than other Daubechies wavelets and, indeed, a host of other orthogonal wavelets [12].

- The research article on ‘Application of Wavelet in Speech Processing of Cochlear Implant’ describes how cochlear Implant bypasses damaged hair cells and stimulates the auditory nerve directly in the cochlea. Wavelets are especially useful for speech processing of CI, since in the case of wavelets one speaks of a constant filter bank as well as the cochlea. A continuous wavelet has application in the filter bank design and an effective result is obtained. Paper also suggests how the lifting scheme, the so-called second generation of wavelet, is adopted, combined with pitch extraction algorithm, to extract fundamental frequency of speech signal [13].

- The technical article on ‘Noise reduction in hearing aids: a review ‘ shows how speech intelligibility for people with hearing loss is affected by background noise. The problem of reducing noise in hearing aids is one of great importance and great difficulty. Paper also describes the problem has been addressed in many different ways over the years. The techniques used range from relatively simple forms of filtering to advanced signal processing methods [14].

- The research paper on ‘An Adaptive Noise Canceller For Hearing Aids Using Two Nearby Microphones ‘ describes the interference of noises and undesirable sounds is very disturbing for speech recognition in normal hearing conditions, especially for hearing-impaired listeners. This is due to reduced hearing abilities that lead, increased masking effects of the target signal. A number of techniques based on single and multimicrophone systems have already been applied to suppress unwanted background noise. Paper is single microphone techniques generally perform poorly when the frequency spectra of the desired and the interfering sounds are similar, and when the spectrum of the interfering sound varies rapidly [15].

- The technical article on ‘A Robust Voice Activity Detection Algorithm in Non stationary Noise’ describes a new voice activity detection (VAD) algorithm to improve the speech detection robustness in nonstationary noisy environments. It shows wiener filtering speech enhancement is adopted to suppress noise from noisy speech. Then, at back-end,
the voice activity detector based on mel filter-bank spectral entropy is presented to
distinguish speech from noise. It has been evaluated that system performance under noisy
environments [16].

- The technical article on ‘Discrete Wavelet Transform Techniques In Speech Processing’
  shows how in many applications of noise cancellation the changes in signal
  characteristics could be quite fast. This requires the utilization of adaptive algorithms,
  which converge rapidly. From this point of view the best choice is the recursive least
  squares (RLS) algorithm. Unfortunately this algorithm has high computational
  complexity and stability problems. Paper also shows an algorithm based on adaptive
  filtering with averaging [17].

- The research paper on ‘Modified LMS Algorithms for Speech Processing with an
  Adaptive Noise Canceller’ describes how a desired signal corrupted by additive noise can
  often be recovered by an adaptive noise canceller using the least mean squares (LMS)
  algorithm. Paper also shows a major disadvantage of the LMS algorithm is its excess
  mean-squared error, or misadjustment, which increases linearly with the desired signal
  power. This leads to degrading performance when the desired signal exhibits large power
  fluctuations and is a serious problem in many speech processing applications. It is
  designed to solve this problem by reducing the size of the steps in the weight update
  equation when the desired signal is strong [18].

- The research paper on ‘A Robust Voice Activity Detection Based on Wavelet Transform’
  shows how voice activity detection is an important step in some speech processing
  systems, such as speech recognition, speech enhancement, noise estimation, speech
  compression etc. In this paper a new voice activity detection algorithm based on wavelet
  transform is studied in which algorithm taking the energy in each sub band, and by two
  methods one can extract feature vector from these values. Experimental results
  demonstrate advantage over different VAD methods [19].

- The research article on ‘Adaptive Wiener Filtering Approach For Speech Enhancement’
  describes that spectral subtraction is the earliest method for enhancing speech degraded
  by additive noise. This technique estimates the spectrum of the clean (noise-free) signal
  by the subtraction of the estimated noise magnitude spectrum from the noisy signal
  magnitude spectrum while keeping the phase spectrum of the noisy signal. The drawback
of this technique is the residual noise. This paper gives idea of wiener algorithm is based on the fact that the vector space of the noisy signal can be decomposed into a signal plus noise subspace and an orthogonal noise subspace in time domain using wiener filter [20].

• The research article on ‘Hearing Aids For The Profoundly Deaf Based On Neural Net Speech Processing’ describes a new speech processing concept for Cochlear Implant systems. It is based on robust feature extraction and a neural net classifier: Feature coefficients, extracted either by relative spectral perceptual linear predictive technique or regular CI-filtering, are classified into ‘auditory related units’. These clusters are closely related to the statistical distribution of the feature coefficients and represent phonetic units. Paper helps for analysis of CI technique in the system [21].

• The research paper on ‘Perceptual Time-Frequency Subtraction Algorithm for Noise Reduction in Hearing Aids’ describes situation of how many people of the community suffering from the sensorineural hearing disorders. In many cases, hearing aids offer the only solution for people suffering from such disorders. In presence of background noise hearing aids are not working properly. A new integrated approach to the design of a digital hearing aid, based on a wavelet transform, as well as a formulation of the temporal and spectral psychoacoustic model of masking. Paper shows the model, the Perceptual Time-Frequency Subtraction algorithm is developed to simulate the masking phenomena and reduce noise in single-input systems [22].

• The technical article on ‘Principles Of Adaptive Noise Canceling’ shows the basic concept of the LMS (Least-Mean-Square) algorithm to develop an adaptive filter that can be used in ANC (Adaptive Noise Cancellation) applications. The method uses a noisy signal as primary input and a reference input that consists of noise correlated in some unknown way with the primary noise. By adaptively filtering and subtracting the reference input from the primary input, the output of the adaptive filter will be the error signal, which acts as a feedback to the adaptive filter. With this setup, the adaptive filter will be able to cancel the noise and obtain a less noisy signal estimate [23].

• The research article on ‘Robust Noise Detection For Speech Detection And Enhancement’ gives idea about the basic requirement of the noise detector to accurately detect periods of stationary noise and not to misclassify speech as noise. The signal is assumed to be noise if its frequency spectrum remains reasonably constant over a
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defined. This structure provides a better performance than a comparison between each of the delayed frame and the current frame. Paper gives idea about feature extraction and threshold level detection during the initial stages of the investigation and overall decision threshold would have to be computed adaptively [24].

- The technical paper on ‘Multichannel Adaptive Noise Reduction In Digital Hearing Aids’ shows how typically the audiologist selects a number of bandpass filters to design a custom binaural equalizer for the hearing impaired subject and cascades the noise reduction algorithm with the equalizer. This paper explores the concept of performing the adaptive noise reduction in individual bands of the equalizer, a multichannel adaptive noise reduction strategy for the hearing impaired [25].

- The research article on ‘Real time background noise cancellation in end user device’ suggests that intolerable noise has always been undesirable in end user communication devices. This is evident for instance during a cricket commentary by the noise of the crowd, or while conversing on a cell phone if the speaker at the other end is on a busy road or any such environment. Proposed paper provides a real time noise cancellation in end user devices using correlation between first primary noisy speech and reference speech in the LMS background [26].

- The technical article on ‘Noise Cancellation Using Parallel Adaptive Filters’ describes use of communications systems in noisy environments such as cellular telephones in automobiles, and radio systems in the cockpits of tanks, planes, and industrial situations requires that measures be taken to improve the quality of the transmitted speech signal to improve its intelligibility by a listener or a speech recognition system. It presents adaptive filter structures based on the noise cancellation principle originally proposed by LMS filter bank concept. Paper shows concept of adaptive noise cancellation requires primary speech signal that contains both noise and speech and a reference signal that contains noise (correlated in some way with the noise in the primary). The adaptive filter is updated by an algorithm such as the LMS algorithm to remove the noise from the primary signal [27].

- The article on ‘Modified Adaptive Filtering Algorithm for Noise Cancellation in Speech Signals’ gives direction how one relationship between the strength of the speech signal and the masking sound is called the signal-to-noise ratio. Ideally, the S/N ratio is greater
than 0dB, indicating that the speech is louder than the noise. Just how much louder the speech needs to be in order to be understood varies with, among other things, the type and spectral content of the masking noise. The most uniformly effective mask is broadband noise. High-frequency noise masks only the consonants, and its effectiveness as a mask decreases as the noise gets louder. But low-frequency noise is a much more effective mask when the noise is louder than the speech signal, and at high sound pressure levels it asks both vowels and consonants [28].

- The research article on ‘Speech Enhancement under Aviation Noise’ describes how the characteristics of the spectrum of different noises can be analyzed; short-time energy in frequency domain of the signals is used to detect voice activity while signal noise ratio (SNR) is very low. Paper also suggests that transition frames are added through forward decision and backward decision and divided into different groups in detail according to the energy by employing a modified method of spectral subtraction [29].

- The technical article on ‘Design and Implementation of an Adaptive FIR Filter Based on Delayed Error LMS Algorithm’ shows how adaptive filtering techniques are widely used in the fields of signal processing and communication such as noise cancellation and speech coding. Adaptive filters usually need real time ability to process signal. The filter can be designed using the digital adaptive finite impulse response filter based on the delayed error least mean square algorithm. The architecture has good hardware utilization efficiency and it can be easily scaled the filter without reducing the throughput rate [30].