CHAPTER 1

INTRODUCTION

The human voice is an important communication tool and any disturbance in the voice quality may have deep implications in an individual's social and professional life. The acoustic analysis is a foremost technique to methods based on the direct examination of vocal folds, which may reduce the frequency of repeated examinations, since such measurements are able to reveal important physiological characteristics of the vocal tract.

In recent years, significant attention has been given to the domain of voice pathology identification and monitoring. In the voice pathology treatment, patients have to frequently visit the doctor for voice therapy. But the patients are waiting for a long time to consult, and are spending a lot of money to find the pathology because the experts have to find the problems in the vocal folds using some endoscopic procedures only. Totally it is an expensive as well as a time-consuming process. Hence such things made the patients discomfort. This situation paves the way for the research in finding an automated tool to identify the voice pathology. The basic purpose of this automated tool is to help the patients for identifying the pathological problems for further progress. Basically, the voice pathology may cause due to the faults in the speech organs, Autism, mental illness, hearing impairment, Paralysis or multiple disabilities. Clinically there is a wide range of guidelines and methods to find the voice pathology and also these methods are subjective and invasive (Mehta, D. D., & Hillman, R. E. 2008). But it is not the case for the automatic tool for the voice pathology.

1.1.Voice Pathology Process

The acoustic analysis for human voice disorder classification using optimization and machine learning techniques is emphasizing the voice pathology process. From this voice pathology process, the Voice pathology is defined as the study of acoustic measures that relate to voice with alterations in a non-invasive way, acoustic measures are primarily used to recognize the patients with a voice disorder. The voice quality detracts from the ability to function and achieve in the society is termed as voice disorder (Petrović-Lazić, M., 2011). This voice pathology process is one of the emerging areas in the field of biomedical engineering. In voice pathology, the voice disorder is confining the vocal quality detracts from the ability to function. Perhaps, the voice disorder is characterized by the abnormal production or absences of vocal quality, pitch, loudness, resonance, and duration, which is inappropriate for an individual's age as well as gender.

In voice pathology, the voice disorder occurs in the human, and also affects the following organs such as **Bronchus, Diaphragm, Larynx, Lungs, Nasal cavity, Oral Cavity, Pharynx, Thorax, and Trachea.** In this case of voice, disorder might occur while any one of the above organs is affected or infected. The disturbance in the voice quality might occur due to the lack of organs coordination. The voice pathology organ is shown in the following Figure 1.1.

Generally, Voice pathology reveal the voice quality by abnormal functional changes with respect to the Bronchus, Diaphragm, Larynx, Lungs, Nasal cavity, Oral Cavity, Pharynx, Thorax, and Trachea in the vocal folds. This type of voice disorder might affect and persons who are more prevalent to voice disorders are, elderly people between the age group of 40-60 years, people with frequent allergies, asthma, cold and sinus infections, persons who are addicted to using tobacco and those who have undergone pathology related surgery and professionals like singers, public speakers, actors, and teachers.

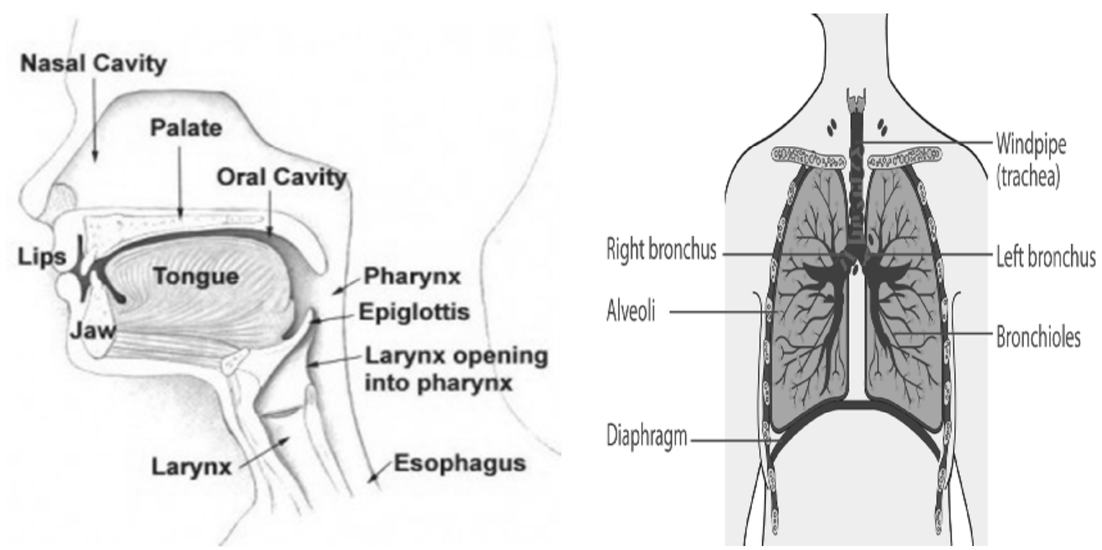

Figure 1.1. Voice Pathology Organs

1.2. Voice Disorders

On being sure of the affected organs, the voice disorder is classified into many types with respect to the symptoms are breathy vocal quality, chronic cough or excessive throat clearing, diplophonic (double-toned) quality, hoarseness, inability to speak loudly, loss of voice, reduced pitch range or sudden change in overall pitch, sudden or gradual change in overall vocal quality, and vocal strain or fatigue. Moreover, the persons are affected by different types of voice disorder, such as Adductor(creates blocked vocal fold), Chorditis (develops inflammation in the vocal cords), Diplophonia (affects the vocal fold, makes it difficult to vibrate), Dysphonia (causes impairment in voice parameters like pitch, intensity and timbre), Keratosis (due to excessive secretion of keratin protein), Laryngitis (due to inflammation in the vocal box), Laryngoceles (the development of air sac, bulge appearance in the neck), Paralysis (because of the weakness in vocal folds), Puberphonia(creates high pitched voice, mostly seen in men), Vocal fold polyp (develops fluid collection on the edge of the vocal cord) and Vocal nodules (develops mass tissue growth in the vocal cord). From these disorders, the Dysphonia, Chorditis, Laryngitis, Laryngoceles, and Diplophonia are considered as major disorder types which are highly prevalent among patients (Pravena, D., & Dhivya, S. 2012).

Especially this research classification focuses on the following voice disorder.

- Glottal Closure Disorder: Consonantal sound used for spoken language will be blocked
- Affected Vocal Cord Stiffness: Voicebox is hardened and fails to produce sound
- Vocal Cord Asymmetry: Lack of equality/equivalence in the organs
- Resonance Chamber Disorder: Quality of speech/prolongation of sound
- Psychological Factors: According to an individual's food habits, environment, and work

From this classification consideration, there are numerous detection techniques to analyze voice pathology detection. This research classifies the 'Pathological' Voice or 'Normal' Voice depending on the successive forward identification analysis such as Preprocessing, Feature Selection, Feature Extraction, and Classification. These successive models can be broadly processed with the input voice signals. The voice input signal data are taken from the private bodies (Department of Pathology, Karpagam Faculty of Medical Sciences and Research, Coimbatore) as a Real-time Dataset and Saarbruecken voice database. These voice databases are tested to detect five main voice Pathological disorders such as **Dysphonia, Chorditis, Laryngitis, Laryngoceles, and Diplophonia.** Finally, the analysis report produces the following consequence.

- Voice disorders can be treated by early identification
- Acoustic analysis is the best support of the diagnosis of vocal and voice diseases
- Acoustic analysis is used to evaluate voice alterations.
- Acoustic analysis is to identify pathological voice signals from normal voice signals.

Investigating the above constraints for voice pathology detection, it has following procedural aspects such as Preprocessing, Feature Selection, Feature Extraction, and classification. This voice database input signal is applied to the Automatic Voice Pathological Identification System (AVPIS). This AVPIS system contains procedural aspects of preprocessing, feature selection, feature extraction, classification, and optimization.

In order to treat voice disorder, early identification and treatment are more important. Acoustic analysis is an efficient tool. By studying the speech signal waves, it is easy to diagnose and screen vocal and voice diseases. Acoustic analysis is used to evaluate the voice alteration in a noninvasive way. One important task is to identify the pathological voice signals from normal voice signals. The pathologist may perform error diagnosis due to tired or overworked and if there is a huge amount of data to examine. Accordingly, error diagnosis is a major drawback for efficient accuracy calculation. To overcome the above drawbacks, this proposed research model is developed to discriminate normal and pathological voices and to carry out automatic screening of voice disorders based on Signal Processing.

1.3 Digital Signal Processing

Digital Signal Processing (DSP) is the numerical control of signals, more often with the aim to Measure, Filter, Produce or Compress continuous Analog Signals. It is described by the utilization of digital signals to represent these signals as Discrete Time, Discrete Frequency, or other Discrete Domain Signals in the form of a sequence of numbers or symbols to allow the digital processing of these signals.

Theoretical investigations and inferences are normally performed on discrete-time signal models, made by the theoretical procedure of sampling. Numerical methods require a digital signal, such as those produced by an Analog-to-Digital Converter (ADC) as shown in Figure 1.2. The processed outcome might be a frequency spectrum or a set of statistics. However it is another digital signal that is converted back to analog form by a Digital-to-Analog Converter (DAC). Regardless of the whole sequence is unpredictable than analog processing and has a discrete value range, the application of computational power to signal processing allows for many advantages over analog processing in many applications, such as error detection and correction in transmission as well as data compression.

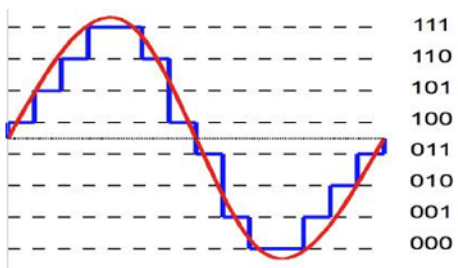

Figure 1.2 Analog to Digital Conversion

Digital Signal Processing and Analog Signal Processing are subfields of Signal Processing. DSP applications incorporate Audio and Speech Signal Processing, Sonar and Radar Signal Processing, Sensor Array Processing, Spectral Estimation, Statistical Signal Processing, Digital Image Processing, Signal Processing for Communications, Control of Systems, Biomedical Signal Processing, and Seismic Data Processing. DSP algorithms have run quite some time on standard computers, as well as on specialized processors called Digital Signal Processors, and on purpose-built hardware such as Application-Specific Integrated Circuit (ASICs).

As of now, there are extra advancements utilized for Digital Signal Processing, including more powerful general purpose Microprocessors, Field-Programmable Gate

Arrays (FPGAs), Digital Signal and Stream Processors (Scalassara P. R., et al 2009). Digital signal processing include linear or nonlinear operations. Nonlinear signal processing is closely related to nonlinear system identification and can be implemented in the Time, Frequency, and Spatio-temporal domains.

1.3.1 Signal Sampling

The expanding utilization of computers has brought about the increased use of, and requirement for, Digital Signal Processing. To digitally examine and manipulate an Analog Signal, it must be digitized with an Analog-to-Digital converter. According to (Dejonckere, P. H., et al 2001), Sampling is usually carried out in two stages, Discretization and Quantization. In the Discretization stage, the space of signals is divided into equivalence classes and quantization is done by replacing the signal with representative signal of the corresponding equivalence class. In the Quantization stage, the representative signal values are approximated by values from a finite set.

The Nyquist–Shannon sampling theorem states that a signal can be exactly reconstructed from its samples if the sampling frequency is greater than twice the highest frequency of the signal, but this requires an infinite number of samples (Costa, S. C., et al 2008). In practice, the sampling frequency is often significantly higher than twice that required by the signal's limited bandwidth. Some continuous-time periodic signals become non-periodic after sampling, and some non-periodic signals become periodic after sampling. In general, for a periodic signal with period T to be periodic with N after sampling with sampling interval T_s, the following must be satisfied given in Equation (1.1)

$$T_S N = Tk \tag{1.1}$$

where *k is an integer*

1.3.2 Domains using Digital Signal Processing

In Digital Signal Processing usually has an investigation of digital signals in the following domains: Time Domain, Spatial Domain, Frequency Domain and Wavelet Domain. They choose the domain in which to process a signal by making an informed assumption as to which domain best represents the essential characteristics of the signal. A sequence of samples from a measuring device produces a temporal or spatial domain representation, whereas a discrete Fourier transform produces the frequency domain information, that is, the frequency spectrum. Autocorrelation is defined as the cross-correlation of the signal with itself over varying intervals of time or space.

a. Time and Space Domains

The most common processing approach in the time or space domain is enhancement of the input signal through a method called filtering. Digital filtering generally consists of some linear transformation of a number of surrounding samples around the current sample of the input or output signal. There are various ways to characterize filters (Benba, A., et al 2014), Such as:

- A "Linear" filter is a linear transformation of input samples; other filters are "non-linear". Linear filters satisfy the superposition condition, i.e. if an input is a weighted linear combination of different signals, the output is a similarly weighted linear combination of the corresponding output signals.
- A "Causal" filter uses only previous samples of the input or output signals; while a "non-causal" filter uses future input samples. A non-causal filter can usually be changed into a causal filter by adding a delay to it.
- A "Time-Invariant" filter has constant properties over time; other filters such as adaptive filters change in time.
- A "Stable" filter produces an output that converges to a constant value with time, or remains bounded within a finite interval. An "unstable" filter can produce an output that grows without bounds, with bounded or even zero input.
- A "Finite Impulse Response" (FIR) filter uses only the input signals, while an "Infinite Impulse Response" filter (IIR) uses both the input signal and previous samples of the output signal. FIR filters are always stable, while IIR filters may be unstable.

A filter can be represented by a block diagram, which can then be used to derive a sample processing algorithm to implement the filter with hardware instructions. The output of a linear digital filter to any given input may be calculated by convolving the input signal with the impulse response.

b. Frequency Domain

Signals are converted from time or space domain to the frequency domain usually through the Fourier transform. The Fourier transform converts the signal information to a magnitude and phase component of each frequency. Often the Fourier transform is converted to the power spectrum, which is the magnitude of each frequency component squared. The most common purpose for analysis of signals in the frequency domain is analysis of signal properties (Maryn, Y., et al 2010). The researcher can study the spectrum to determine which frequencies are present in the input signal and which are missing.

In addition to frequency information, phase information is often needed. This can be obtained from the Fourier transform. With some applications, how the phase varies with frequency can be a significant consideration. Filtering, particularly in non-real time work can also be achieved by converting to the frequency domain, applying the filter and then converting back to the time domain. This is a fast, O (n log n) operation, and can give essentially any filter shape including excellent approximations to brick wall filters (Saenz-Lechon, N., et al 2006).There are some commonly used frequency domain transformations. For example, the Cepstrum converts a signal to the frequency domain through Fourier transform, takes the logarithm, then applies another Fourier transform. This emphasizes the harmonic structure of the original spectrum. Frequency domain analysis is also called Spectrum or Spectral Analysis.

c. Z-Plane Analysis

Whereas analog filters are usually analyzed in terms of transfer functions in the s plane using Laplace transforms, digital filters are analyzed in the z plane in terms of Z-transforms (Huang, P. W., & Lee, C. H. 2009). A digital filter may be described in the z plane by its characteristic collection of zeroes and poles. The z plane provides a means for mapping digital frequency to real and imaginary z components using Equation 1.2.

$$\text{where} \quad \mathbf{Z = re^{jw}} \quad \text{for continuous periodic signals and}$$

$$\mathbf{w=2\pi F} \text{ (} \mathbf{F} \text{ is the digital frequency).} \qquad (1.2)$$

This is useful for providing a visualization of the frequency response of a digital system or signal.

d. Discrete Wavelet Transformation

In numerical analysis and functional analysis, a Discrete Wavelet Transform (DWT) is any Wavelet Transform for which the wavelets are discretely sampled. As with other wavelet transforms, a key advantage it has over Fourier transforms is temporal resolution: it captures both frequency and location information (Carvalho, R. T. S., et al 2011).

1.3.3 Applications

The main applications of DSP are audio signal processing, audio compression, digital image processing, compression, speech, speech recognition, digital communications, radar, sonar, financial signal processing, seismology and biomedicine. Specific examples are speech compression and transmission in digital mobile phones, room correction of sound in hi-fi and sound reinforcement applications, weather forecasting, economic forecasting, seismic data processing, analysis and control of industrial processes, medical imaging such as CAT scans and MRI, MP3 compression, graphics image, Hi-Fi loud speaker crossovers and equalization, and audio effects for use with electric guitar amplifiers.

a. Application of Digital Signal Processing in Voice Disorder Diagnosis

The generation of vocal sounds is generally referred to as “phonation,” while the action of generating word sounds is referred to as “speech” or “articulation.” The organs involved in phonation and/or speech are the oral cavity, nasal cavity, pharynx, larynx, trachea, bronchus, lungs, thorax, and diaphragm; these multiple organs work in a coordinated manner to perform complex integrated movements (Fonseca, E.S. and Pereira, J.C., 2009).

Awan, S. N., & Roy, N. (2009). proposed Signals mainly deal with extraction of features from voice signals and combined with the pattern classification methods provides the expert systems for the detection of voice pathologies shown in Figure 1.3.

In the phonation mechanism, in response to a command to vocalize from the cerebral cortex, the respiratory muscles contract and expiratory flow from the lungs is moved upwards towards the trachea and larynx as a power source, while at the same time both vocal cords are adducted through the recurrent laryngeal nerve, closing the glottis.

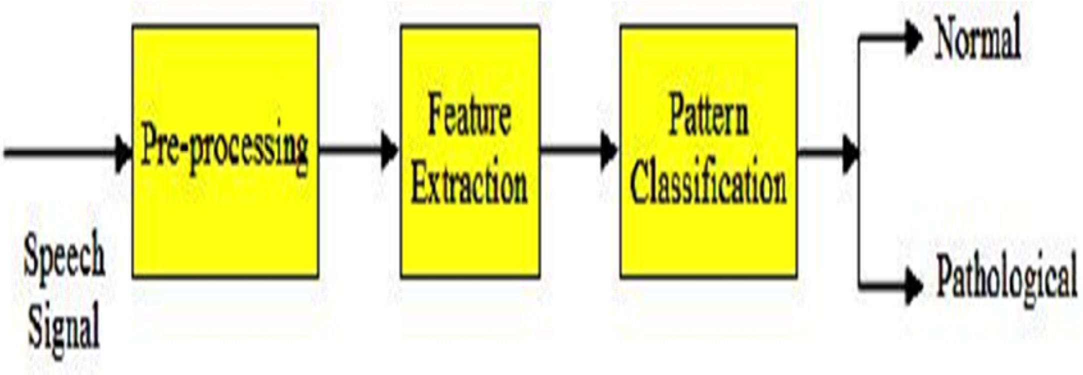

Figure 1.3 Voice Disorder level based on Speech Signal

The expiratory flow raises the sub glottal pressure, causing the vocal cords to vibrate and generate sound, which passes through the vocal tract (which acts as a resonance chamber) to produce vocal sound. Consonants and vowels are articulated, becoming speech, and are generated continuously to produce spoken words (Eadie, T. L., & Doyle, P. C., 2005). If any part of these phonation control or motion mechanisms becomes impaired, a voice disorder occurs.1) The clinical state of voice disorders can be classified from phonation mechanisms as follows: 1) glottal closure disorder; 2) affected vocal cord stiffness; 3) vocal cord asymmetry; 4) respiration/ resonance chamber disorder; and 5) psychological factors of these, 1) to 3) is abnormalities in the shape and motility of the larynx and are the main causes of voice disorders. When examining patients who are complaining of voice disorders, first of all they are asked about their main complaints, medical history, degree and quality of hoarseness, past history, occupation, and daily lifestyle habits and social background related to phonation, and possible causes of the voice disorder are estimated.

A physician who is a voice specialist can estimate the patient’s condition just by listening to their voice. Furthermore, performing indirect laryngoscope or laryngeal endoscopy enables the diagnosis of many laryngeal disorders (Fex, B., Fex, S., et al 1994). Initial responses to patients by general practitioners differ according to whether the disorder they have is benign or malignant, acute or chronic.

Although it is thought that patients with benign disorders such as acute Corditis Vocalis with a common cold are in many cases initially treated at internal medicine clinics, in cases where there is a high degree of hoarseness and the hoarseness has not improved in two weeks or more, the patient is referred to a physician specializing in ear, nose, and throat disorders. Loosely examining a patient without looking at the larynx can result in malignancies such as laryngeal cancer and thyroid cancer being overlooked. Also, patients with acute epiglottitis or other airway stenotic disorders should be referred to a specialist physician urgently.

b. Diagnosis for Voice Disorders

With regard to the patient's chief complaints, they are asked about how phonation has been impaired. It is important to obtain a present illness including the time since onset of the symptoms and treatment at other hospitals (Fredouille, C., et al 2005). With regard to contributory factors, the patient is asked about voice misuse, past operations on the larynx, and past operations for which the patient was under general anaesthesia. If the patient has experienced symptoms of heartburn, acid reflux, or reflux esophagitis in the past, the possibility of laryngeal granuloma can be considered. If the patient complains of respiratory difficulties, there is the possibility of airway stenosis and the patient is referred to a specialist. With regard to occupation, misuse the voice and vocal cord nodules are easily formed in cases with vocal abuse (Hadjitodorov, S., &Mitev, P. 2002). Dysphonia plicae ventricularis can be observed amongst Buddhist monks. For restaurant and service industry employees, smoking, drinking, and karaoke singing also exert an effect, with vocal cord polyps and polypoid vocal cords occurring more commonly.

c. Inspection and Palpation

With regard to inspection of the oral cavity and oropharynx, the clinical features of acute inflammation, such as mucosal reddening or adhesion of purulent mucus, are observed. With regard to palpation of the neck, for acute disorders the presence/absence and location of tenderness is checked, and for cases in which a malignant disorder is suspected, special attention is paid to cervical lymphadenopathy and thyroid tumours.

d. Maximum Phonation Time

For Maximum Phonation Time (MPT), the maximum length of time a patient can vocalize after taking a deep breath is measured. In general, 10 seconds or less is abnormal, and 5 seconds or less interferes with daily living. Disorders which shorten MPT include recurrent nerve paralysis and vocal cord atrophy.

E. Auditory-Perceptual Evaluation of Hoarseness

An auditory-perceptual evaluation method for hoarseness is the GRBAS scale of the Japan Society of Logopaedics and Phoniatrics, which gives scores of 0, 1, 2, or 3 for the Grade of hoarseness; Roughness, Breathiness, Asthenia, and Strain, where 0 is normal, 1 is a slight degree, 2 is a medium degree, and 3 is a high degree (Gerratt, B. R., et al 1993). Rough hoarseness is a rasping, rattling sounding voice that can be heard mainly in disorders such as vocal cord polyps, polypoid vocal cords, and laryngeal cancer. Breathy hoarseness is a whispery voice that can be heard in such disorders as recurrent nerve paralysis, vocal cord nodules, laryngeal cancer, acute Corditis Vocalis, and vocal cord atrophy. Asthenic hoarseness is a small, weak voice which is heard in such disorders Aspsychosomaticaphonia and myasthenia gravis. Strained hoarseness is produced with the throat constricted; this condition occurs in such disorders as spasmodic voice disorders and laryngeal cancer. In addition, disorders in which the voice becomes muffled include Peritonsillar abscess and acute epiglottitis; these are potentially lethal disorders which cause airway stenosis and must not be overlooked (Kasuya, H., et al 1986).

1.4 Problem Statement

To design and develop an Automatic Objective Assessment for diagnosis of voices based on Acoustic Analysis for Human Voice Disorder Classification Using Optimization and Machine Learning Techniques.

1.5 Motivation of the Research

- This research considered the design and development of an automatic objective assessment for diagnosis of voices based on signal processing.

- The lab testing is less effective in voice diagnosis; this also is of higher costs in terms of diagnosis time and economic status.
- Pathology and normal patients are giving as the less reliable participation in the quality in the diagnosis of voices system.
- Taking care of a difficult issue is less on the analysis of voice process with additional time.
- Identification between pathology and normal detection of disease is less productive.

To overcome this process, the automatic voice pathological identification system is developed in this research.

1.6 Objective

The objective of this research work is to classify the voice signals using acoustic factors such as Signal Energy, Pitch, Silence Removal, Noise Removal, Mel Filtering. The classification techniques namely Support Vector Machine and BPNN algorithm are employed to optimize the predicted disordered voice signal as Normal voices and Pathology voices by Modified Optimized Back Propagation Network Disorder Voice Classification (MOBPNDVC). The pitch modulation constant used to recognize the pathology voice and normal voices as male or female. This analysis is used to find and resolve the pathological clinical diagnosis of the disordered voice signal as male and female.

1.6.1 Specific Objectives

- To study voice pathology characteristics of “disordered voice signal” motivates us to improve and classify pathological clinical diagnosis by classifying the “input voice signal” as ‘Pathological’ or ‘Normal’ voices,
- To study pathological clinical diagnosis properties of ‘Pathological’ or ‘Normal’ voices are identified with respect to the Pitch from the noisy disordered voice signal,
- To avoid the curse dimensionality and selection of relevant features,
- To improve accuracy with feature extraction process like LPC, MFCC with Cat Swarm Optimization technique,

- To classify the voice disorders using SVM algorithm & BPNN,
- To enhance the operations of each step of the automatic system in order to improve the overall performance,
- To compare Specificity, Sensitivity, Execution Time and Accuracy analysis are vital a domain in pathological clinical diagnosis subjects, and
- To find the accuracy of pathology voice from Input voice signal is fast and leading efficient, this motivates to recommend this system to pathological clinical sector.

1.7. Chapter Formulation

The underlying objective of this research work is to design and develop an Automatic Objective Assessment for diagnosis of voices into normal or pathological. The documentation is organized as follows:

C**hapter 1** provided the introduction to voice pathology process and its disorders. Also, the problem statement was stated with respect to the voice pathology process. The motivation of the research should be encouraging for efficient pathology process. Moreover, the objectives of the study must convey the research contribution and its application.

Chapter 2 review the previous research contribution of various researchers' ideas is discussed and compare the research with the proposed research contribution. The recent research concepts are discussed with respect to the Pathological Voices analysis. The research related to the Pathological Voices and Parametric Representation by preprocessing are also discussed. The research related to the Feature Selection and Extraction is also discussed. Classifications of Voiced Data of Pathological Voices research article were also discussed. The result and discussion about the pathology diagnostic method are compared with the other method and researcher ideas.

The Research Design procedures of the Automatic Voice Pathological Identification System are discussed in this chapter, for understanding the research methods in a simple manner. Finally, the Research Design process was drawn in **Chapter 3**

Chapter 4 explores the parametric representation by preprocessing. The Preprocessing and Preprocessing Research Design are also discussed. The functional

Requirement for the Preprocessing is discussed on the Saarbruecken Voice Database (2000 persons) and private Real-time Dataset (80 samples). This chapter also discussed the Acoustic Parameters and Acoustic analysis for Noise Removal, signal energy, Silence removal, pitch, Electro Glotto Graph (EGG). The Wiener Filter and DWT Filters discussed for making the hybrid wiener filter discrete wavelet transforms for denoising estimation and SNR analytical examination values with respect to the Signal to Noise ratio Calculation

The Signal Energy, Pitch and Making the structure of extracted affected voice signal and Application of algorithm involvement of affected voice signal. Also, the Feature Selection Process, Automated Voicing Analysis (AVA), Dimensionality Reduction were discussed. The Pathology Thresholding is discussed for the Feature Extraction Process. The proposed Cat Swarm Optimization Mel Frequency Cepstrum Coefficients (CSOMFCC) with Linear Predictive Coding (LPC) is also discussed. The Validation of Features are done by using the Cat Swarm Optimization (CSO) are also discussed. Finally, the (CSOMFCC) area conferred in **Chapter 5**

Chapter 6 analyzes the Proposed Classification of Voiced Data with respect to the Support Vector Machine (SVM). The Back Propagation Learning Algorithm (BPA) are also discussed for the formation of proposed Modified Optimized Back Propagation Network Disorder voice Classification (MOBPNDVC).

The Results and Discussions of the proposed system with various pathology classification methods and performance metrics with the help of ROC curve is deliberated in **Chapter 7**. The work carried out in the research work along with future research directions is summarized in **Chapter 8**

The work of several researchers is quoted and used as evidence to support the concepts explained in this thesis. All such evidence used is listed in the reference section of this thesis.

1.8. Research Contribution

- Proposed the Hybrid Wiener Filter Discrete Wavelet Transforms (HWFDWT) for Silence Removal and Noise Removal in Pre-processing techniques. It outperforms well in denoising the voice signals.
- Proposed the Wavelet Thresholding Algorithm for Silence Removal and Noise Removal.
- Proposed Cat Swarm Optimization Mel Frequency Cepstrum Coefficients (CSOMFCC) to extract the best features from the signal in the Feature Extraction process.
- Proposed the Modified Optimized Back Propagation Network Disorder Voice Classification (MOBPNDVC) for Voice Pathology Identification System construction.
- Proposed MOBPNDVC Optimization Algorithm for Disorder Voice classification accuracy calculation.
- Analyzed the Voice Pathological Identification System to identify five types of voice pathology such as Laryngitis, Laryngoceles, Dysphonia, Diplophonia, and Chorditis.

1.9. Chapter Summary

The underlying objective of this research work is to develop an Automatic Voice Pathological Identification System using optimization and machine learning techniques. Techniques that improve the existing methods in various stages of the development of the system are proposed. For this purpose, the existing techniques were studied carefully in order to build an enhanced system which will identify the disorders in the speech signals. The results of such a literature study are presented in the next Chapter, Chapter 2, **Review of Literature**.