

Signal Processing Techniques to Estimate the Speech Disability in Children

Anandthirtha. B. Gudi, H.K. Shreedhar and H. C. Nagaraj

Abstract— Speech disability is one of the widely encountered disabilities among the disabled children. In this study, adaptive signal processing is used to tune the speech signal of disabled children into curve fitted speech signal of normal child by implementing feedback mechanism technique. The curve fitted speech signal constants of normal child is compared to tuned constant values of disabled child and the deviation is calculated using correlation technique. The amount of deviation represents the severity of speech disability in children compared to a normal child. The speech therapy can be given to the speech disabled children to improve the utterance capability.

Index Terms— Adaptive signal, Correlation, Filter Bank, Spectrogram, Speech disability

I. INTRODUCTION

Speech is one of the very common means of communication between humans in any civilized society. In speech communication an idea is first formed in the mind of speaker and then it is transformed into words, phrases and sentences according to the grammatical rules of the language. However in our society, many people are suffering from disabilities related to speech communication. Speech disability forms one of the widely encountered among the various disabilities among children [18-6]. Various techniques for speech reproduction were researched by several researchers over the years like articulation density index [2], Hidden Markov model [9-22], Hidden Bernoulli model [14]. Extract the vocal parameters from recorded speech sound for the vocal disorder, speech assessments and some neurological pathologies identification [7]. Examine both excitation and vocal tract characteristics for overall pathology assessment using a hidden Markov model (HHM) recognizer [8]. Acoustic signal analysis using artificial neural network technique in the task of recognition of selected types of vocal tract pathologies [31]. This research work is mainly

focused on the speech disability severity observed in children of age between five to eight years (pathological subjects) and measurement of such speech disability severity in comparison with the normal fit children (normal subjects) of the same age group.

In this paper the speech uttered by children of age group five to eight years was recorded and digitized. The digitized signal was further processed in high level computational platform MATLAB. In section II the procedure of speech recording and digitization were explained; In section III spectrogram has been discussed; section IV deals with curve fitting procedure for speech signals; section V deals with adaptive signal processing technique; section VI deals with creation of an image using speech signal samples; VII deals with speech signal processing through the filter bank; in section VIII obtained results are discussed and conclusions were drawn in section IX. This study is a new technique and unique in its kind to assess the speech disability. The flowchart presented as figure 1 describes the present work.

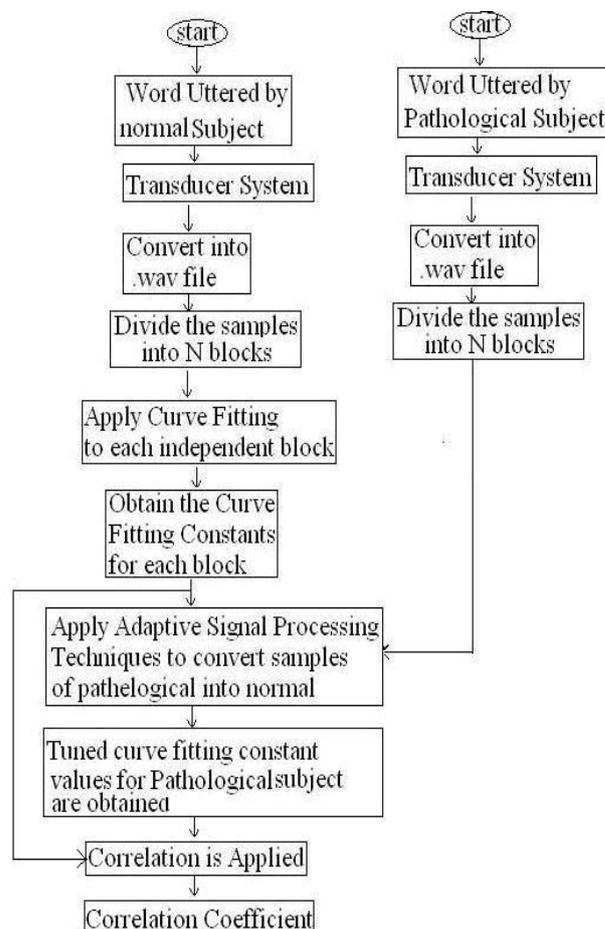


Figure 1. Methodology to obtain correlation between the normal and pathological subjects

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II. SPEECH RECORDING PROCEDURE

Speech signals are continuous signals in time domain. To convert such signals into discrete time signals, Analog to Digital (A to D) converters are used. Current state of the art recording devices such as Intelligent Portable Occular Device (IPOD) has built in advanced A to D converter. This device is used for the creation of speech signal at discrete intervals of time.

The study was conducted on children of native speakers of Kannada language (Indian Dravidian Language which has recognized as classical language recently) Acoustic speech signal of word “Namma” uttered by normal subject and pathological subjects(age between five to eight years) are recorded using IPOD recorder in digital form.

The data was collected at J.S.S. ‘Sahana’ integrated and special school located at Bangalore, India. Prior to the data collection, the teachers were requested to train and acclimatize the disabled children of age between five to eight years facing speech disability problem, to read few sentences written in Kannada language. The experiment was conducted in a pleasant atmosphere, the children are made to read the sentences and the said sentences were recorded. Similarly the normal samples were obtained at ‘vinayaka’ school, Bangalore by following the same procedure with the normal children in the same age group. About 30 samples were taken in all for normal cases. The averages of these samples were taken and the child’s voice which nearly coincides with the average is considered as reference data mentioned as ‘normal’. However, the data after conversion should be made easy to use, manipulate, low loss and compatible to windows platform. The recorded digitized signal is further subjected to transformation into wav file, the most commonly used digitized sound format with the help of downloadable software known as “GOLDWAVE”. The transformation procedure used in GOLDWAVE software results in .wav file format. Table-1 shows the parameters of .wav file obtained.

TABLE I. THE PARAMETERS USED FOR TRANSFORMATION INTO .WAV FILE

Rate[kbps]	512
Amplitude type[bit]	16
Sampling Frequency[Hz]	32000
Channel	Mono
Modulation	Pulse Coded

III. SPECTROGRAM

A spectrogram is an image that shows how the spectral density of a signal varies with time. The spectral intensity at each point in time is indicated by the intensity (darkness) of the plot at a particular analysis frequency. During the periods of unvoiced speech high-frequency energy is seen in the spectrograms; during periods of silence no spectral activity (because of the reduced signal level) is seen.

Creating a spectrogram using the STFT is usually a digital process. Digitally sampled data, in the time domain, is broken up into blocks, which usually overlap, and Fourier transformed to calculate the magnitude of the frequency

spectrum for each block. Each block then corresponds to a vertical line in the image; a measurement of magnitude versus frequency for a specific moment in time. The spectrums or time plots are then "laid side by side" to form the image or a three-dimensional surface. The spectrogram is given by the squared magnitude of the STFT of the function.

Spectrogram has been obtained for the word “Namma” uttered by normal subject, pathological subject1, pathological subject2 and pathological subject3. Such spectrograms are shown in figure 2 (a), 2(b), 2(c) and 2(d) respectively.

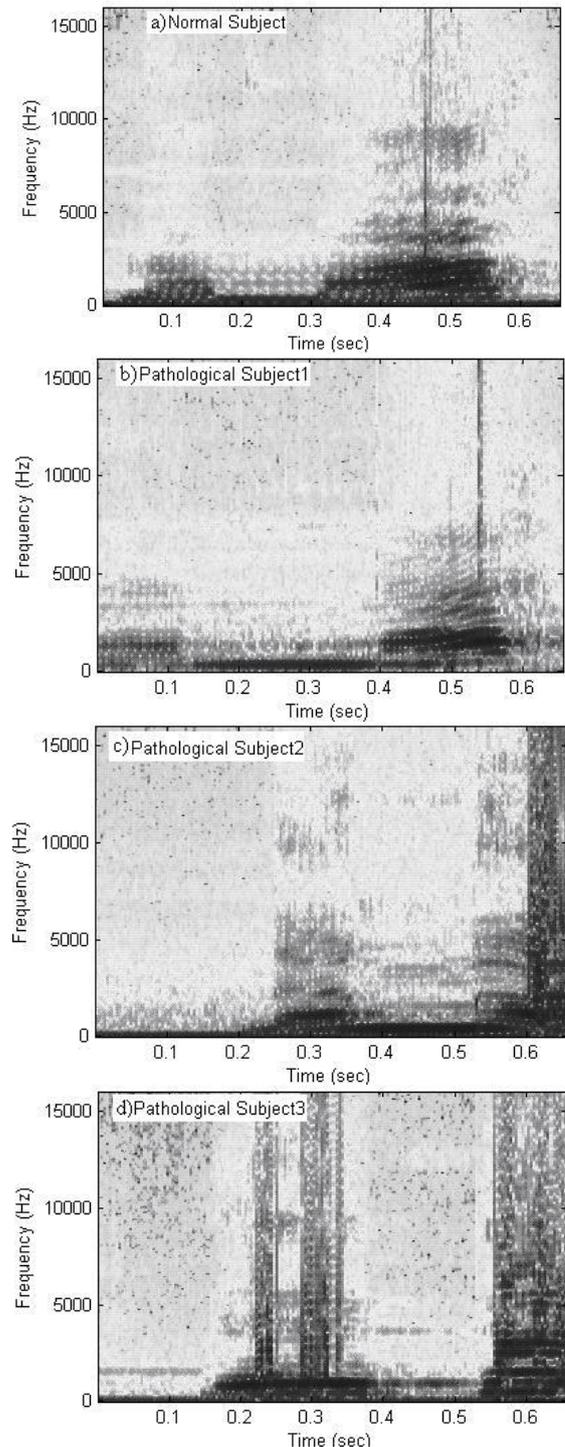


Figure 2. a) Spectrogram of Normal Subject b) Spectrogram of pathological Subject1 c) Spectrogram of pathological Subject2 and d) Spectrogram of pathological Subject3

By observing the above spectrograms it is seen that the

spectrogram of pathological subject 1 is closer to that of normal subject with less distortion. Spectrogram of pathological subject 2 is seen to be more deviated from that of normal subject with moderate distortion. Spectrogram of pathological subject 3 is seen to be most deviated and highly distorted from that of normal subject.

From the above we can conclude that the speech disability severity is mild with pathological subject 1, moderate with pathological subject 2 and severe with pathological subject 3 in comparison with normal subject.

IV. CURVE FITTING PROCEDURE

Recorded acoustic speech signals for Kannada word "Namma" uttered by the normal subject is taken and MATLAB curve fitting tool (cftool) is used to achieve the better curve fitting for the same. To have better curve fitting with confidence level of more than 90% and with a minimum value of Root Mean Squared Error (RMSE), the sample sequence set is divided in to few number of windows. Each window must consist of some suitable number of samples.

The curve fit techniques such as Gaussian, Polynomial, Interpolent Cubic Spline, Rational, and Sum of Sine Functions have been applied in the present study and it was found that the sum of sine function technique give the best curve fit for speech signals along with curve fit coefficients. From so obtained curve fit coefficients mathematical equation can be derived for the speech signal. Though the Interpolent Cubic Spline technique yields better curve fit for speech signal curve fit coefficients can not be obtained. Hence the sum of sine function technique was adopted for curve fitting in this study. This is clearly evinced in the present study. The mathematical function (sum of sine functions) given in the equation (1) is used to get proper curve fitting with the above said window samples.

$$a_1 \sin(b_1 X_{nj} + c_1) + a_2 \sin(b_2 X_{nj} + c_2) + \dots + a_8 \sin(b_8 X_{nj} + c_8) \quad (1)$$

a_1 to a_8 , b_1 to b_8 , and c_1 to c_8 are constants, X_{nj} represent the sample sequence, where $j=1, 2, 3 \dots$

V. ADAPTIVE SIGNAL PROCESSING

An adaptive system is the one, whose structure is alterable in such a way that the performance improves through contact with its environment. In other words, the main property of the adaptive system is its time – varying and self – adjusting performance. Adaptability is a main property of most of the biological systems. More recently adaptive signal processing algorithms are being implemented. The aim of all adaptive signal processing algorithms is to reduce the error as small as possible. This has to be done by adapting the input signal such that the system output is very similar to the desired signal. The required information pertaining to the difference of actual signal and the required signal is transferred into a set of adjustable system coefficients. Following are the some of the application examples of adaptive signal processing. a) System identification b) Inverse system identification c) Prediction d) Interference cancellation.

The “sum of sine function” can be rewritten to suite for different windows of pathological subject as equation (2) as shown below.

$$a_1 \sin(q_1 jX_{pj} + c_1) + a_2 \sin(q_2 jX_{pj} + c_2) + \dots + a_8 \sin(q_8 jX_{pj} + c_8) \quad (2)$$

X_{pj} represents the speech signal samples sequence set of pathological subject where $j=1, 2, 3$

To adapt the pathological subject’s sample values into that of normal subject, equation (1) and (2) are equated.

$$a_1 \sin(b_1 X_{nj} + c_1) + a_2 \sin(b_2 X_{nj} + c_2) + \dots + a_8 \sin(b_8 X_{nj} + c_8) = a_1 \sin(q_1 jX_{pj} + c_1) + a_2 \sin(q_2 jX_{pj} + c_2) \dots + a_8 \sin(q_8 jX_{pj} + c_8) \quad (3)$$

From the Equation (3) it can be assumed that

$$a_1 \sin(b_1 X_{nj} + c_1) = a_1 \sin(q_1 jX_{pj} + c_1) \\ a_2 \sin(b_2 X_{nj} + c_2) = a_2 \sin(q_2 jX_{pj} + c_2) \\ \dots a_8 \sin(b_8 X_{nj} + c_8) = a_8 \sin(q_8 jX_{pj} + c_8)$$

The values q_{1j} , q_{2j} , q_{3j} , q_{8j} can be tuned as follows:

$q_{1j} = b_1 X_{nj} / X_{pj}$, $q_{2j} = b_2 X_{nj} / X_{pj}$, $q_{8j} = b_8 X_{nj} / X_{pj}$
 If the pathological sample value is zero, then it is equated to unity to avoid the error in the division process

The block diagram of the above said adaptive process is shown in figure 3

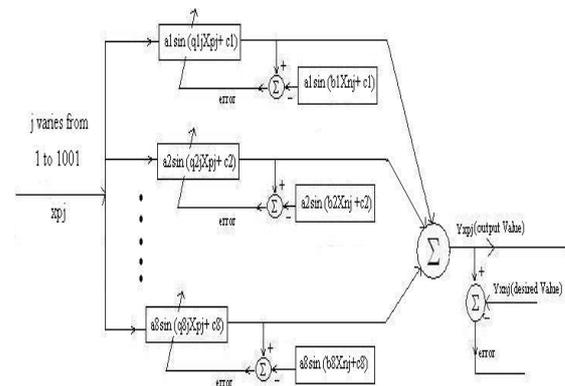


Figure 3. Adaptive process to tune the utterance sample values of pathological subject into normal

Curve fitting constants b values of normal subject for the uttered word “Namma” are correlated with tuned q values obtained for the pathological subject after the adaptive process. Such correlated values obtained with three different pathological subjects in comparison with normal are tabulated in table 2.

TABLE II. CORRELATION STUDY

Sl. No.	Comparative Study	Correlation Coefficient
1	'normal and subject 1'	0.0078
2	'normal and subject2'	0.0039
3	'normal and subject3'	0.0032

VI. CREATION OF AN IMAGE USING SPEECH SIGNAL SAMPLES

Total numbers of 21100 speech signal samples obtained for the uttered word "Namma" are squared and logarithm to the base 10 is applied to each of them. Then matrix of dimension 100 X 211 has been created. This matrix has been utilized to create the image. This procedure is applied for all the subjects. Obtained diagram in each case is shown in figure4

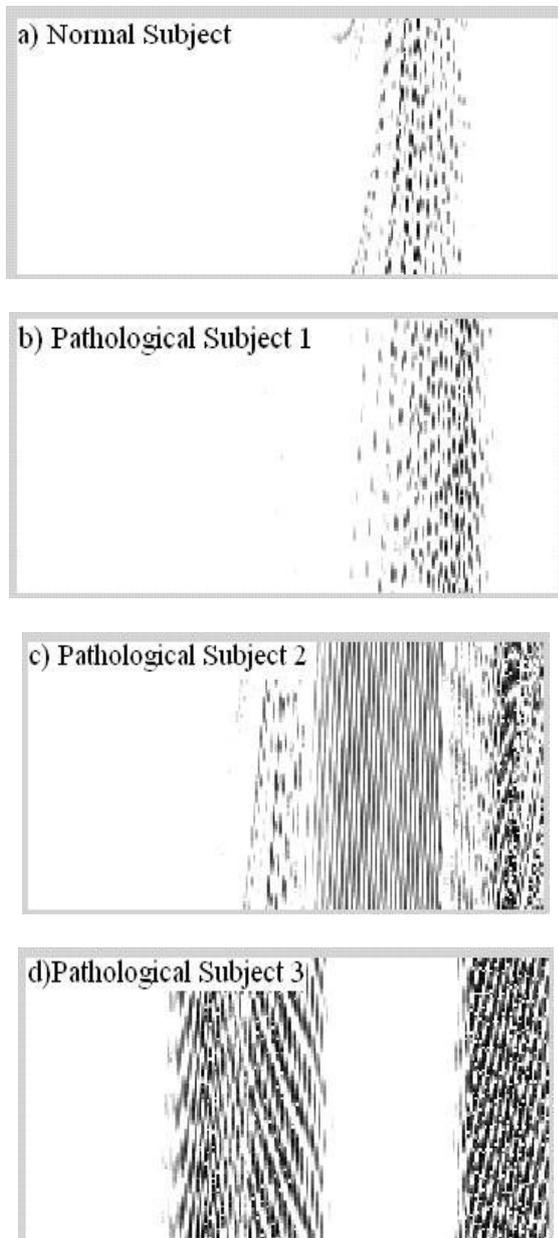


Figure 4. Image obtained using speech signal samples of

a) Normal subject b) Pathological subject 1 c) Pathological subject 2 and d) Pathological subject 3

By visualizing the above images following observations are drawn.

- 1) Image of pathological subject 1 is seen to be closer to image of normal subject.
- 2) Image of pathological subject 2 is seen to be moderately deviated from that of normal subject.
- 3) Image of pathological subject 3 is seen to be heavily deviated from image of normal subject.

From the above we can conclude that the speech disability severity is mild with image of pathological subject 1, moderate with image of pathological subject 2 and severe with image of pathological subject 3 in comparison with normal subject.

VII. SPEECH SIGNAL PROCESS THROUGH THE FILTER BANK

A Finite Impulse Response (FIR) filter with band limit between 30Hz to 3400Hz is designed and speech signal samples of normal subject are passed through the filter. The same procedure is employed to obtain individual sample sequence for each pathological subject. Then correlation coefficient value is calculated between the absolute values of filter out put obtained for normal subject and that of pathological subject. The results are tabulated in Table III.

TABLE III. CORRELATION COEFFICIENT OF DIFFERENT CASES

Bandpass Filter Band width.	Comparative Study Between	Correlation coefficient
30Hz – 3400Hz	Normal child and pathological subject-1	0.394
	Normal child and pathological subject-2	-0.0473
	Normal child and pathological subject-3	-0.2643

FIR filter bank is obtained by four FIR band pass filters designed with Hamming Window. The pass band limits of each filter is tabulated in Table IV.

TABLE IV. FILTER BANK DETAILS

Filter No.	Lower cutoff frequency	Upper cutoff frequency
FIR Band pass filter-1	30 Hz	900 Hz
FIR Band pass filter-2	900 Hz	1800 Hz
FIR Band pass filter-3	1800 Hz	2700 Hz
FIR Band pass filter-4	2700 Hz	3400 Hz

Speech signal samples of normal subject are passed through the filter bank. The same procedure is employed to obtain individual sample sequence for each pathological subject. Then correlation coefficient value is calculated between the absolute values of individual filter out put obtained for normal subject input and that of pathological subject. The results are tabulated in Table V.

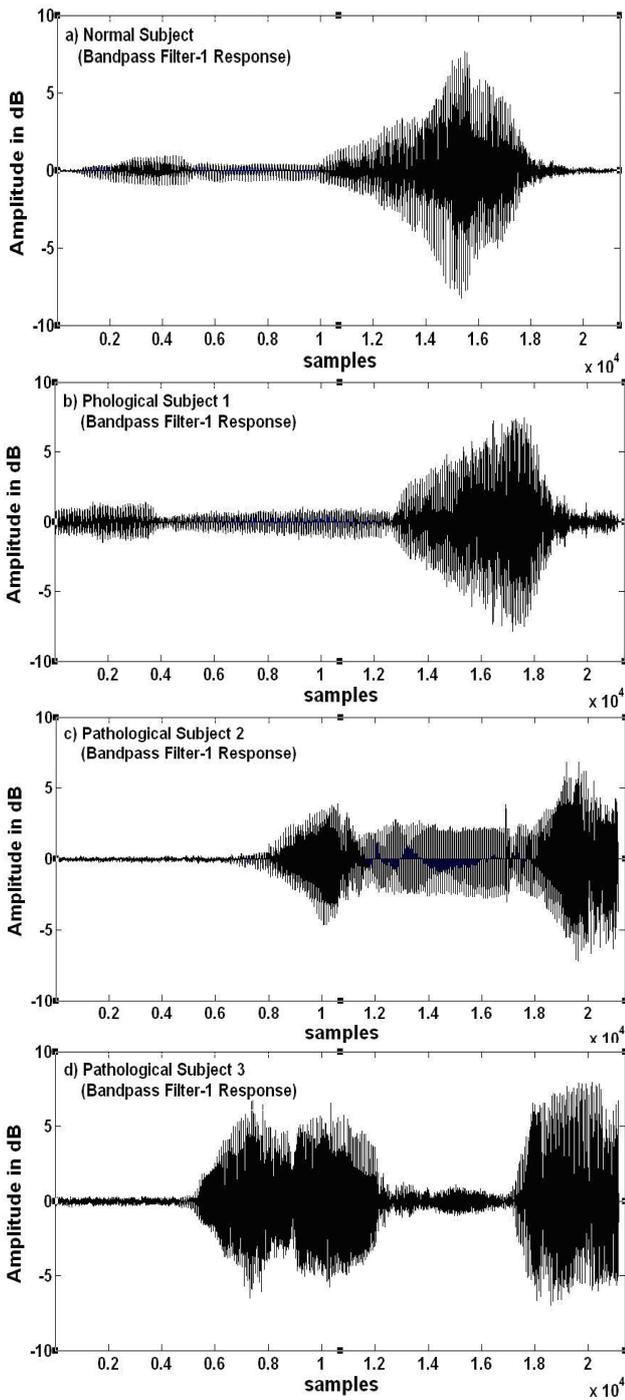


Figure 5 Band pass filter-1 response for a) Normal subject speech sample sequence b) Pathological subject-1 speech sample sequence c) Pathological subject-2 speech sample sequence and d) Pathological subject-3 speech sample sequence.

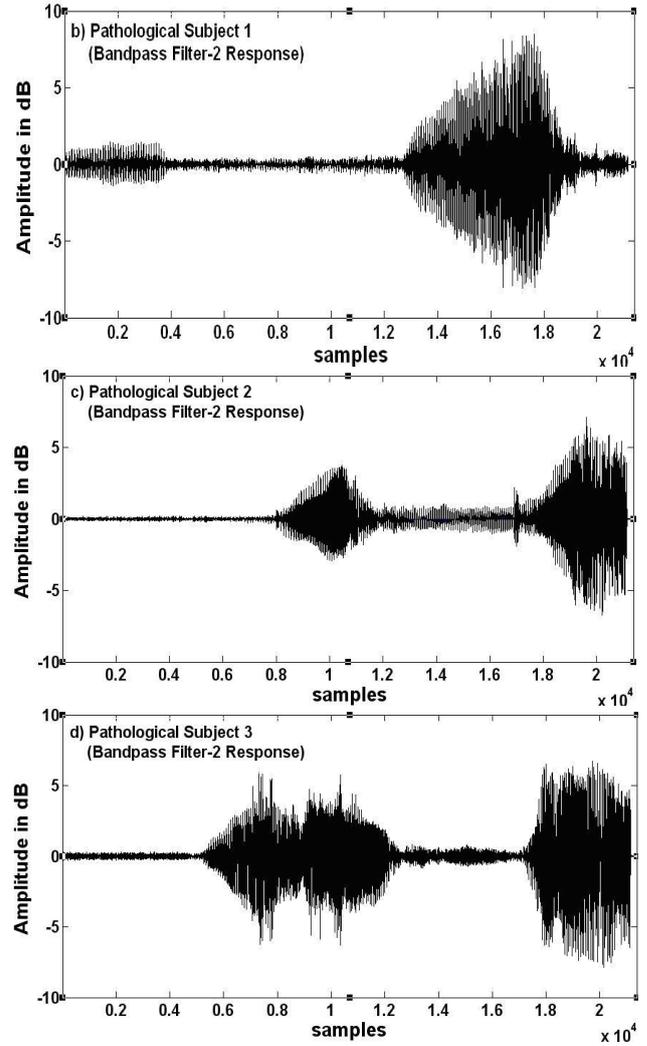
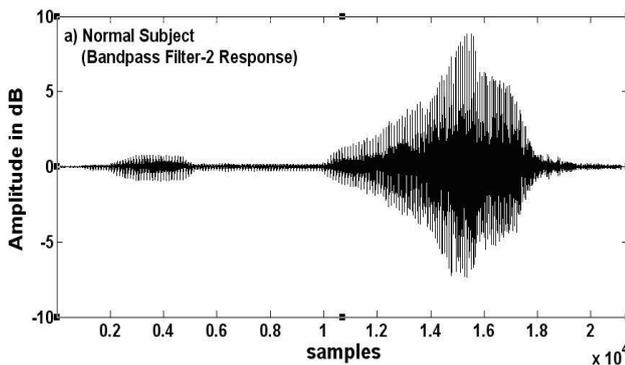
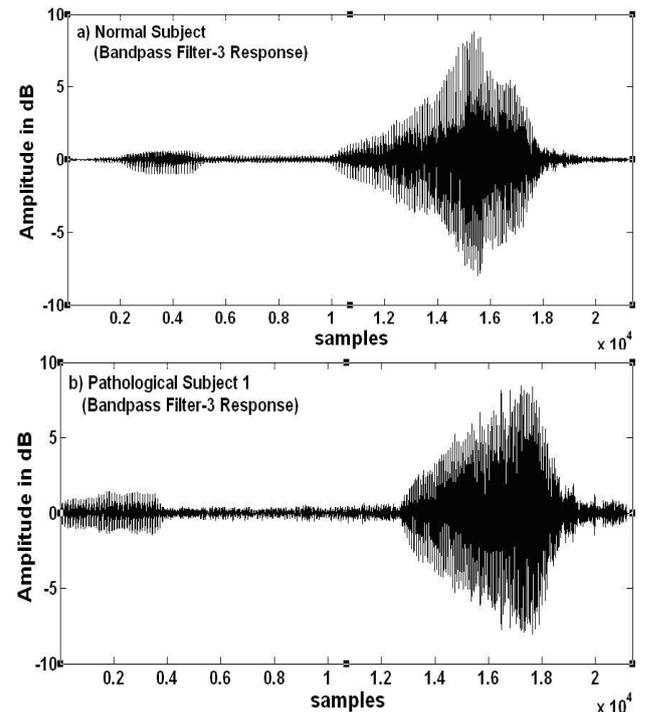


Figure 6 Band pass filter-2 response for a) Normal subject speech sample sequence b) Pathological subject-1 speech sample sequence c) Pathological subject-2 speech sample sequence and d) Pathological subject-3 speech sample sequence.



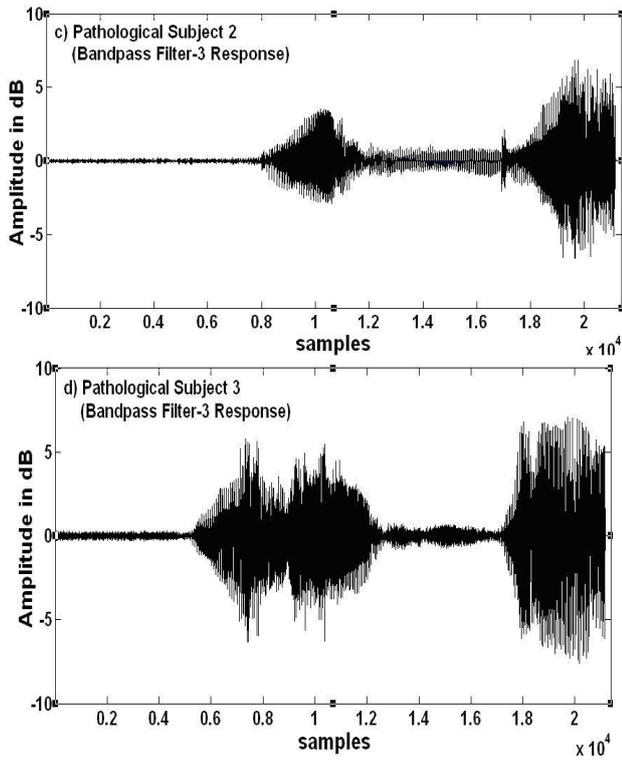


Figure 7 Band pass filter-3 response for a) Normal subject speech sample sequence b) Pathological subject-1 speech sample sequence c) Pathological subject-2 speech sample sequence and d) Pathological subject-3 speech sample sequence.

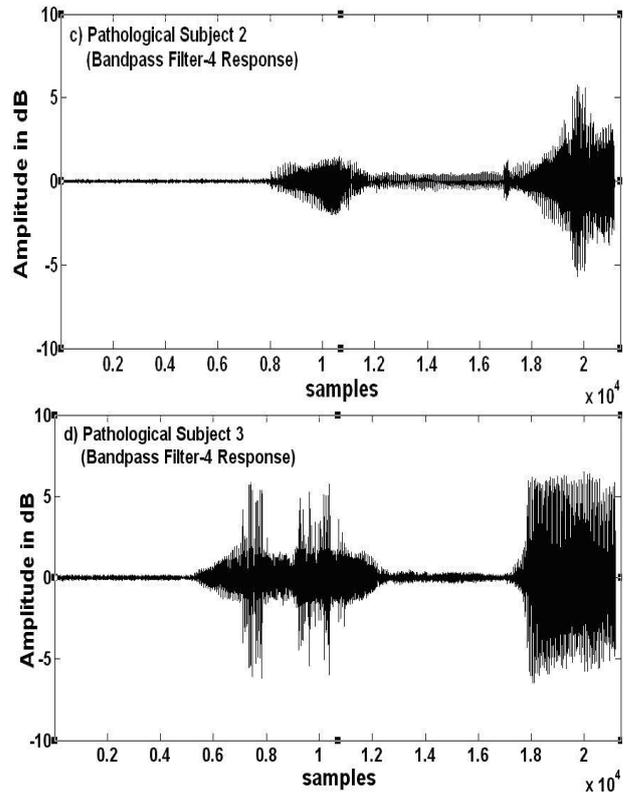
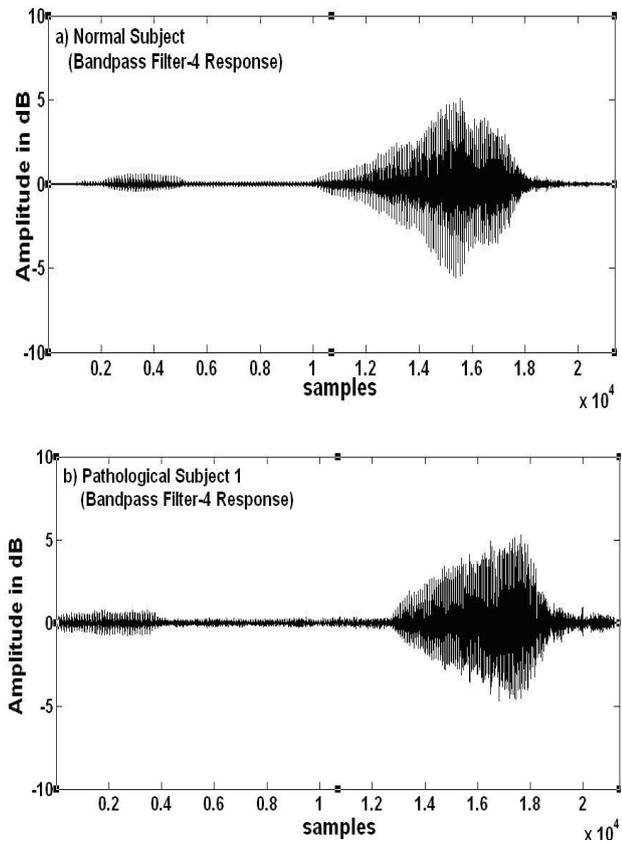


Figure 8 Band pass filter-4 response for a) Normal subject speech sample sequence b) Pathological subject-1 speech sample sequence c) Pathological subject-2 speech sample sequence and d) Pathological subject-3 speech sample sequence



Correlation coefficient values obtained with filter bank is more effective and speech disability severity measure can be done very effectively compared to a single bandpass filter. With filter bank method also, the speech disability severity observed is mild, moderate and severe in pathological subject 1, subject2 and subject 3 respectively in comparison with normal subject.

TABLE V. CORRELATION COEFFICIENT OF DIFFERENT CASES USING FILTER BANK

Comparative Study Between	Bandpass Filter Number and Band width	Correlation Coefficient
Normal child and pathological subject-1	Bandpass Filter-1 30Hz – 900Hz	0.5263
	Bandpass Filter-2 900Hz – 1800Hz	0.5523
	Bandpass Filter-3 1800Hz – 2700Hz	0.5521
	Band pass Filter-4 2700Hz –	0.5602

Normal child and pathological subject-2	Band pass	0.0753
	Filter-1	
	30Hz – 900Hz	
	Band pass	-0.123
Normal child and pathological subject-3	Filter-2	
	900Hz – 1800Hz	
	Band pass	-0.124
	Filter-3	
Normal child and pathological subject-3	1800Hz – 2700Hz	
	Band pass	-0.129
	Filter-4	
	2700Hz – 3400Hz	
Normal child and pathological subject-3	Band pass	-0.290
	Filter-1	
	30Hz – 900Hz	
	Band pass	-0.295
Normal child and pathological subject-3	Filter-2	
	900Hz – 1800Hz	
	Band pass	-0.294
	Filter-3	
Normal child and pathological subject-3	1800Hz – 2700Hz	
	Band pass	-0.263
	Filter-4	
	2700Hz – 3400Hz	

VIII. DISCUSSION

By increasing the number of filters in filter bank with in the specified frequency band width (30Hz to 3400Hz) assigned for the filter bank, speech disability severity can be measured more accurately.

If correlation coefficient is a positive value and close to unity then speech disability will be mild and if correlation coefficient is small positive value or small negative value then speech disability will be moderate. If correlation coefficient is more negative, then speech disability will be very severe.

From the tabulated results in Table-2, Table-3 and Table-5 we can conclude, Pathological subject-1 is having mild speech disability, Pathological subject-2 is having moderate speech disability and Pathological subject-3 is having severe speech disability.

IX. CONCLUSION

This study helps the speech therapist to distinguish the pathological subjects suffering from speech disability from normal subjects. Adaptive signal processing techniques lead to better accuracy and precision in assessing the speech disability of a subject. This also helps in knowing the level of severity of speech disability.

APPENDIX

MATLAB programme to create an image/Pattern using

speech signal samples

```
clear;
clc;
figure;
```

```
x=wavread('Normal Subject.wav');
%Start point and End point of the speech
%Segment for the uttered word "Namma" is
%given below
y=x(1302720:1323819);
yy=y.*y;
y1=-log10(yy);
z1=reshape(y1,100,211);
subplot(2,2,1);
imshow(z1);
xlabel('Normal Subject');
```

```
x1=wavread('Pathological Subject1.wav');
%Start point and End point of the speech
%Segment for the uttered word "Namma" is
%given below
y2=x1(353280:374379);
yy1=y2.*y2;
y3=-log10(yy1);
z2=reshape(y3,100,211);
subplot(2,2,2);
imshow(z2);
xlabel('Pathological Subject1');
```

```
x2=wavread('Pathological Subject2.wav');
%Start point and End point of the speech
%Segment for the uttered word "Namma" is
%given below
y4=x2(613120:634219);
yy2=y4.*y4;
y5=-log10(yy2);
z3=reshape(y5,100,211);
subplot(2,2,3);
imshow(z3);
xlabel('Pathological Subject2');
```

```
x3=wavread('Pathological Subject3.wav');
%Start point and End point of the speech
%Segment for the uttered word "Namma" is
%given below
y6=x3(1116800:1137899);
yy3=y6.*y6;
y7=-log10(yy3);
z4=reshape(y7,100,211);
subplot(2,2,4);
imshow(z4);
xlabel('Pathological Subject3');
```

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