Journal Homepage: -www.journalijar.com
INTERNATIONAL JOURNAL OF ADVANCED RESEARCH (IJAR)
ISSN NO. 2320-5407
Article DOI:10.21474/IJAR01/ 9257
DOI URL: http://dx.doi.org/10.21474/IJAR01/9257

RESEARCH ARTICLE

GENDER IDENTIFICATION USING FUZZY LOGIC AND NEURAL NETWORK MODEL.

K. Manikandan[1] and Dr. E. Chandra[2].
1. Department of Computer Science, PSG College of Arts and Science, Coimbatore, India.
2. Professor and Head of the Department, Department of Computer Science, Bharathiar University, Coimbatore, India.

……………………………………………………………………………………………………....

Manuscript Info

……………………….

Manuscript History
Received: 12 April 2019
Final Accepted: 14 May 2019
Published: June 2019

Key words:-
Neural Network, Fuzzy logic, Fast Fourier Transform.

Abstract

………………………………………………………………….

Speech investigation is Art of the most exciting fields in Digital Signal Processing system. The reason only most of the researches have been made on it under the different resources tools and logical programs to generate an analysis that begin from speech production, speech recognition, speech processing, speech coding. F.J .Taylor is the initially to apply the analysis of speech signal [1]. In this research we handle the, males and females speech specimen of the utterance, 'Close', be used to make a system in Fuzzy Logic and then Neural Network to identify the male as of female words tone and similarity of between the results of the both systems, then the fuzzy logic approach was improving based on three features of the spokesman tone which are the signal value of the energy rate, the signal of the power spectrum and vowel sound “letter O” in the word close. In this speech example to enlarge to ability in recognizing an individual speaker and then improve the system protection against intruders by building the system recognizes the vocalizations of a individual person register a voice approval authority of that person to make an access denied to unauthorized user to avoid accessing the system. The system shows good results during testing operation implement samples of one person next to others person like males and females samples**.**

……………………………………………………………………………………………………....

Introduction:-

The human speech signal is a highly outmoded and non-stationary signal. This parameter occur the speech signal to be very complicated to work in the ASR [2]. The SR systems depends on the following two main system is[3]

1. Speaker dependent systems.
2. Speaker independent systems.

In broadly, (SR)speaker recognition can be under taken by speaker identification and speaker verification .In further the speaker identification can be closed-set else open-set and also, each deviation is to implemented as text-depended otherwise text-independent [4]. human being speech tone can be divided into two distinct sections: one for sound production and another one for sound shaping. Sound production is denote by air fleeting across the vocal cords ("a", "e", and "o") or from a structure in the vocal tract ("sss", "p", or "sh"). Sound production with help of

Corresponding Author:-K. Manikandan.
Address:-Department of Computer Science, PSG College of Arts and Science, Coimbatore, India.

vocal chords has been taken as a feature parameter is tracking the vowel sound for testing speech in male and female samples in my research; unvoiced speech is produced by the following attributes , nasal passages
lips, teeth, and mouth , tongue. Sound shaping is a mixing of the vocal tract, and unvoiced speech attribute. In speech Recognition, sound shaping is called filtering [3]. MATLAB series was processing in this research and it's easy to implement non-stationary signals [5].

Resources and Technique

Recording the Speech Samples

In this paper we first to collect the both females and males sample voice and also individual authorized voice speaker sample for a female. the special word for collecting the speaker's voices was saying,' Close', word as the vocal cords audio can be heard perfectly.

Low-Pass Filter

It is mostly important tool for speech recognition" low-pass filter" because the recording voice is suppose noise back ground means a major information is low frequency rate so this filter was applying twice in following approach , when reading the speech signal in MATLAB program environment and then when calculating the power of the DFT to speech signal from the noise. Fig(1).

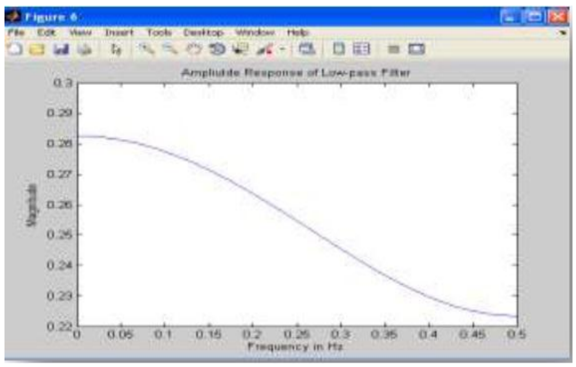

Figure 1:-The Effect of the Low-Pass Filter on the Power of a sample of the Speech Signal

Features extraction

Pitch

Pitch is the main deviation between male and female speakers. A human speech pitches from in the vocal cords. The motive of pitch against between sexes is the ratio and the tension of laryngeal trade and in which includes the vocal glottis.The primary frequency of human voice is 250 Hz(length 10.4mm) .The vocal fold length increase males and females like 15-25mm and 13-15mm. The range of female's pitch is highest then average pitch can be found between 120 and 200 Hz. Because female the size of their larynx is smaller [4]. So that this feature was mostly useful in recognizing between the female and the male voice in many working research in this area. We take in this way of pitch feature we initiate amplitudes of the power of voice signal is one more feature that can be used to find out the persons voice By implementing the Fast Fourier transform method.

Fast Fourier Transform (FFT)

The discrete Fourier transform (DFT) is the main role of digital signal processing and image processing area. The DFT is broadly used as features extractor because the frequency is holding the information about the pitch and the formant. The DFT of the n vector length x is another vector y of length n.

$$X(k) = \sum_{j=1}^{N} x(j) W_N^{(j-1)(k-1)}$$

$$x(j) = (1/N) \sum_{k=1}^{N} X(k) W_N^{-(j-1)(k-1)}$$

The transform signal was help to analyze the power of the DFT for the speech signal ratio .The result was shown in the plot of periodogram as shown in Fig (2).

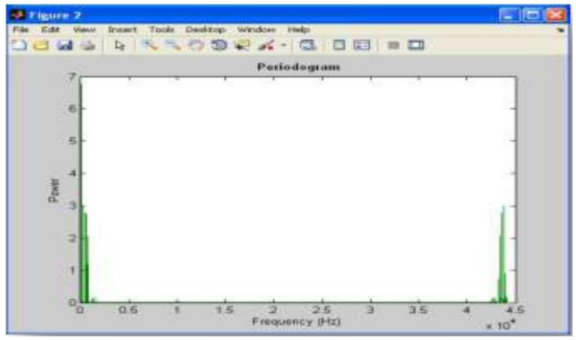

Figure 2:-The Periodogram Plot of the speech signal

C. Spectrogram

Spectrogram is one of important technique for frequency depiction of the original speech. In this technique is allows users to make out the amount of energy of speech. This is most important tool to identify voiced and unvoiced areas. In the research spectrogram tool .We used in MATALB language because the frequency of power spectrum speech signals noise cancellation and system identification. The process started in MATLAB is loading a speech signal from MATLAB workplace for the word,' Close', for each sample. Then sorting out the speech signal to n number of frames with particular time (1/8000) and 80 samples per frame. The buffering of I/O is 128 samples per frame were to reduce the estimating noise. After that Fourier transform was implementing and so for the signal using Periodogramblocks the result is calculate a nonparametric ratio of the power spectrum of speech signal.

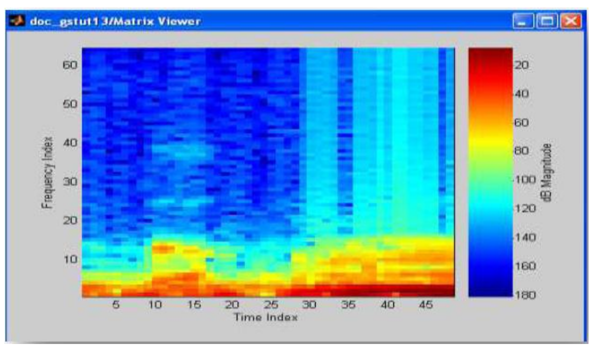

Figure 3:-Matrix Viewer Window displaying the Spectrogram of the Speech Signal

From Fig (3), we can detect the harmonics speech signal when the vowel, "o" is spoken in the word, "Close", where mainly this speech signal's energy is determined. The Spectrograms stand for the color-based visualizations of the development of power spectrum of a speech signal it is calculated using periodogram power spectrum estimation method.

Training Neural Networks For Speech Recognition

It is a computational system inspired by the Structure, Processing Method,and Learning Ability of a biological brain. A large numberof very simple processing and weighted interconnection between neuron elements. The well-known illustration of a NN training algorithm is called as back propagation and conjugate gradient descent are significantly faster for many problems, in case of BPN network still has more useful in some conditions, and also understanding algorithm is very simple. In back propagation, the gradient vector is pointing the error surface is calculated. The complicated part is to know how large the steps must be. Large steps may meet more quickly than simple steps, but may also exceed the solution or terminate in the incorrect direction.

A typical model of this in NN training is somewhere the algorithm progress very limitation of along a steep, narrow, valley, bouncing from one part from others in contrast, very simple steps involve in the correct direction, but they need to a more number of iterations. The accurate learning ratio is application-dependent, and is classically implement by experiment. The algorithm so for progresses flow of through a amount of epochs. On each epoch, the working out cases is individually submitted in the line of network, and destination and actual outputs is matched and the error ratio calculated. This error rate is mutually among the error surface gradient, is help to change the weights, and then the process again and again. The initial network configuration is arbitrary and sampling is terminated; when a set of epochs elapse else when the fault reaches an suitable level, or when the error end rate is improving. In the research, BPN Network has been applied to make a network that must capable of recognize male from female's samples with help of the power amplitudes reaction of the discrete Fourier transform for the speech signals training, where the output shows that the power amplitude answer for males' training voice were greater than the females' sample voice amplitude response.

1. Specify the phonetic categories that the network will be make out .In the research we implement the power amplitudes for the DFT for both males and females samples.
2. In this paper the perfect male and female testing scenario based on its power amplitudes is used. The higher power value for male sample is assigned to the target vector [0 0 1 1] .the lower power value for the female sample is assigned to target vector [0 1 0 1] .
3. Initially the Training a neural network to recognize males and females testing voice. In this paper the Back Propagation Neural Network was educated for 362 epochs, by way of 50-hidden unit as a launch but it's not satisfy the number of recognition process so for the hidden units are increased 50 to 100-hidden unit to improve the results.
4. To analyze the network working principles using a test set. In this paper we used the network was tested using 55 – sample voice for females' and males' attribute for using MATLAB neural network tool environment. Fig (4) neural network training state.
5.

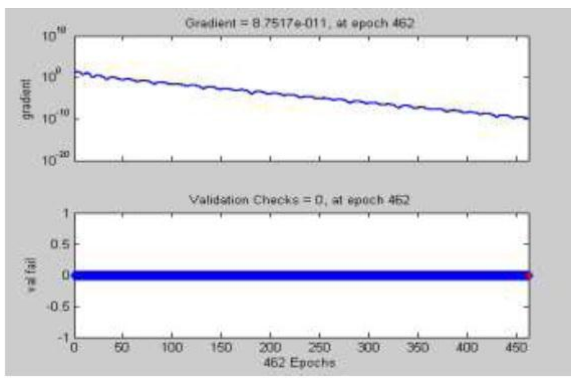

Figure4:-The Training State of the Back propagation Neural Network

Fuzzy Logic

Fuzzy sets are sets whose elements have degrees of membership. Fuzzy sets were introduced by Lotfi A. Zadehand Dieter Klauan 1965 .Fuzzy sets are a super –set of classical sets .In classical set theory the fundamental membership of a set is used in binary terms by two valued logic condition. Fuzzy sets are usually articulated as a set of elements that having degree of membership for the truth values in closed unit interval [0, 1]. The idea behind a fuzzy set represents a concept and having a context is a further expanded by linguistic variables. A linguistic variable is assigned to a fuzzy region, a set of fuzzy sets that represent a complete concept. And it is also a super-set of classical logic that deals with prepositions which required being either 1or 0. And it rule is well with noise data. These personality as in [10] recommended in the researches that FL system might be an successful tool for speech recognition. The MATLAB fuzzy logic toolbox more useful tool for solving problems in speech recognition. It helps to generate and correct fuzzy inference systems (FIS) by using graphical tools and command-line functioning [7]. In this paper initially first fuzzy system approach was to make a recognize between females and males samples is on two rules; If the power amplitude of the speech signal is small value then female speaks and large value then male speaks. In this two rules is useful to recognize the speaker as male and as a female; the system is apply that mamdani MATLAB toolbox. In this tool box under

1. the extracted features,
2. power-amplitude,
3. power spectrum and
4. vowel-sound,

The above attributes was make to be able to recognize individual speaker of a female samples next to other females and males training sample. In this approach was to use of 24- rules under the values of extracted features, with three inputs denotes the 3- extracted features for training samples, help to the power amplitude of the DFT, The power Spectrogram and signal energy are the most important feature that was determined in the vowel sound 'o' of a, "Close" word. This above features was able to recover the required vocalizations signal for the required speaker. The system is shown in Fig (5)

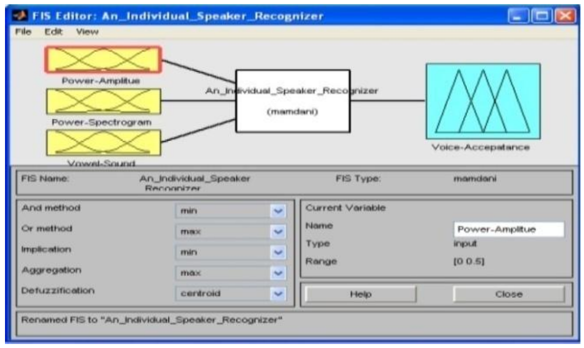

Figure 5:-The Individual Speech Signal Recognizer

Experiment Results:-

First we collecting the speech samples for male's and female's both of samples, the BPNN was experienced to recognize the speech signal with help of 50-hidden units, The target was well thought-out [0 0 1 1] for dissimilar males-M- sample and dissimilar [0 1 0 1] for females-F-samples .The output of training [0 -0.8779e- 011 1 1; 0 1 -3.1255e-011 1],

Recorded Speech	**Female Speech Output [0 1 0 1]**
F1	0 0.8779 0.8736 1
F2	0 1.7524 0.0828 1
F3	0 1.1320 0.6414 1
F4	0 1.8760 -0.0714 1
F5	0 0.523 -3.2951 1
F6	0 1.0743 0.1981 1
F7	0 1.4856 0.0081 1
F8	0 -0.0337 1.0192 1
F9	0 0.2674 1.0682 1
F10	0 -0.091 2.7629 1

Recorded Speech	**Male Speech Output [0 0 1 1]**
M1	0 -0.4529 3.0165 1
M2	0 1.5731 0.2354 1
M3	0 0.1836 0.9397 1
M4	0 1.7054 0.1696 1
M5	0 0.1481 0.9373 1
M6	0 -0.2679 1.6234 1
M7	0 1.8879 0.0667 1
M8	0 0.5963 1.0559 1
M9	0 -0.4214 3.4166 1
M10	0 1.6542 0.3876 1

Table1:-The Testing Results of the BPNN with 50-Hidden Units

The Back propagation neural network was trained with need to help of the male and female samples for particular epochs as applied before, the speech energy values extracted feature in the speech proposition, the resulting values was declared as [0 0 1 1] for male speech and [0 1 0 1] for females speech after that the BPNN was tested against return to the same operation is performed in specific criteria in BPNN. The results are around particular values as get the above tables; the BPNN was trained and experienced using 50-hidden units, and also 100 units. Table (2) is the recognition result using BPNN for a variety of numbers of hidden units. We mentioned the table (2) hidden units value is increased; the presentation of the BPNN was enlarged True samples rate, while the (False) samples did not produce the required correct results.

Recognition Success (50-hidden units)	
Female (True)	50%
Female (False)	70%
male (True)	70%
male (False)	50%
Recognition Success (100-hidden units)	
Female (True)	44.7%
Female (False)	75.6%
male (True)	83.4%
male (False)	17%

Table 2:-Recognition result

The FIS was built using MATLAB tool to recognizer speech system for males and females samples. Two rules were used in the system to recognize the male from the female samples as mentioned above.

If the speech signal the power amplitude is small value then female speaks. (0 – 0.5).

If the speech signal the power amplitude is large value then male speaks. (0 – 0.5).
The output rate for the speaker was (0 – 10), female speech rate value (0 – 2.5), while male speaker is (2.6 and above).

Recognition Success (Females		**Recognition Success (Males)**	
True %	90 %	True %	85.6 %
False %	10 %	False %	23.5 %

Table 3:-Speech Recognizer Fuzzy Interface System

But this approach was not satisfied clearly to recognize an every speaker to be tested adjacent to others speakers. 10-Samples speech is collected for an individual speaker all fit in to one female record then tested beside on the other sample speech. Speech sample acceptance is based on two case ,First case is Access-Validate in range 0-5 speaker authority to access a job area or protection system , second case to produce Access- not Validate in range 6-10 to give a rejection of authorized voice , The three extracted features signal process are discussed in table (4).

Power Amplitude range	(0 – 0.5) watt.
Power Spectrogram range	(-100 – 0) db
Vowel Sound,’O’	(- 100 – 0) db.

Table 4:-Signal process rate

MATLAB handling slab set and the harmonics vowel sound letter, ’O’, word ‘Close’, the first fuzzy logic system to recognize the speech sample based on two rules is called the amplitude value ,so for its very difficult for the system to recognize the required speakers using BPNN so improve the fuzzy logic approach gives more adaptability in speech processing is based on 24- rules shaped a robust fuzzy recognizing speaker sample. Table (5); we mention the victory rate for recognizing the essential speaker

Recognition Success (Individual Speaker)	
True %	98.1%
False%	02.8 %

Table 5:-Recognizing speaker sample

Discussion:-

Speech study is silent tough area in recognition, classification and retrieval .several researchers have finished in speech study, in this paper we proved Fuzzy logic was successful system individual speaker recognition ,retrieval system , compared with Back Propagation neural network . Fuzzy logic is a comfort system in speech retrieval because true, false (1,0) decisions based on a membership functions. The first system is recognition rate is difficult one so in this paper we increase the recognition capability of this system with help of spectrogram analysis and also increase the system security. the fuzzy interface system(FIS) based on 24- rules a choice was complete to give a voice approval to the speakercalled “individual Speaker Recognizer”,,"Access- Validate, Access- Denied ",to other unauthorized speakers. The results of table (5) we mention the recognition accurate rates of this approach. In Future we use this fuzzy logic approach to implement the Gender classification and speaker age identification in neural network area.

References:-

1. Chester , and M.Doyle ,"The Winger Distribution In Speech Processing ", Ho, Octavian, C., and K. Lai, ,” Speech Processing ", prentice Hall ,p 392-p442
2. Jere, B. ," Digital Signal Processing Application using the ADSP-2100 Family ", Application Engineering staff of Analog Devices DSP Division Edited ,PTR ,Prentice Hall ,New Jersey ,p.330-p.331.
3. Brain, J. Love, and Jennifer, V. and, Xuening, S.,"Automatic Speaker Recognition Neural Networks", Electrical and Engineering Department, the University of Texas at Austin, Spring 2004.
4. Abdul-Bary, R., S., Soad, T. A., "MATLAB-Based Design and Implementation of Time-Frequency Analyzer", College of Electrical Eng., Technical Institute.
5. M.Liu, L.Wang," Content-Based Audio Classification and Retrieval using a Fuzzy Logic System towards Multimedia Search Engines".
6. The Math Work, Inc. "MATLAB the Language of Technical Computing ", Version .7.6.0(R2008a), USA.
7. Christos, S. and, Dimitrios, S., "Neural Networks", http://www.emsl.pnl.gov:2080/docs/cie/neural/neural.homepage.html
8. John-Paul, H., and Ron, C. and Mark, F. and John , S. , and Yonghong ,Y. and Wei ,W. , "Training Neural Networks for Speech Recognition", Center for spoken language understanding ,(CSLU), Organ Graduate Institute of Science and Technology,January29,1998.
9. Patrick.M. Mills,"Fuzzy Speech Recognition", Bachelor of Science in Engineering, Swathmore College, 1994, University of South Carolina, Thesis 1996.
10. Dunia, J.,Jamma, "Low Cost Wavel et Analyzer", Computer Engineering Department, Technical College/ Mosul, Foundation of Technical College,Thesis,January/2005.