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Speech Recognition of Isolated Arabic words via using Wavelet Transformation and Fuzzy Neural Network

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Abstract

In this paper two new methods for feature extraction are presented for speech recognition the first method use a combination of linear predictive coding technique(LPC) and skewness equation. The second one(WLPCC) use a combination of linear predictive coding technique(LPC), discrete wavelet transform(DWT), and cpestrum analysis. The objective of this method is to enhance the performance of the proposed method by introducing more features from the signal. Neural Network(NN) and Neuro-Fuzzy Network are used in the proposed methods for classification. Test result show that the WLPCC method in the process of features extraction, and the neuro fuzzy network in the classification process had highest recognition rate for both the trained and non trained data. The proposed system has been built using MATLAB software and the data involve ten isolated Arabic words that are (الله، محمد، خديجة، ياسين، يتكلم، الشارقة، لندن، يسار، يمين، أحزان), for fifteen male speakers. The recognition rate of trained data is (97.8%) and non-trained data is (81.1%).

Keywords: Speech Recognition, Feature Extraction, Linear Predictive Coding (LPC),Neural Network, Fuzzy network

1. Introduction

Arabic, a Semitic language and one of the six official languages of the United Nations (UN), is one of the most widely spoken languages in the world. Statistics show that it is the first language (mother-tongue) of 206 million native speakers ranked as fourth after Mandarin, Spanish and English. In spite of its importance, research effort on Arabic Automatic Speech Recognition (ASR) is unfortunately still inadequate. Modern Standard Arabic (MSA) is the formal linguistic standard of Arabic language, which is widely taught in schools and universities, and used in the office and the media. It has been the focus and the core interest of many previous and recent researches compared to dialectal Arabic (Abushariah2012).

Speech recognition system is divided into two parts, feature extraction and classification. Feature extraction method plays a vital role in speech recognition task. Isolated word/sentence recognition requires the extraction of features from the recorded utterances followed by a training phase. The most widely used feature extraction techniques are the Perceptual Linear Predictive (PLP), the Linear Prediction Coefficients (LPC), the Linear Prediction Cepstral Coefficients (LPCC), Mel Frequency Cepstral Coefficients (MFCC) and various forms of the Mel Frequency Cepstral Coefficients (ΔMFCC, ΔΔMFCC) (Abdalla2013).

In this paper, a new methods for speaker recognition are presented. The first method is based on the use of a combination between linear predictive coding technique(LPC) and skewness equation. The second one(WLPCC) use a combination of linear predictive coding technique(LPC), discrete wavelet transform(DWT), and cpestrum analysis. The objective of these methods are to enhance the performance of the proposed method by introducing more features from the signal. Neural Network(NN) and Neuro-Fuzzy Network are used in the proposed methods for classification.

2. Previous Work

In this paper presents a system for speaker independent speech recognition, which is tested on isolated words from three oriental languages, i.e., Urdu,Persian, and Pashto. The proposed approach combines discrete wavelet transform (DWT) and feed-forward artificial neural network (FFANN) for the purpose of speech recognition. DWT is used for feature extraction and the FFANN is utilized for the classification purpose. The task of isolated word recognition is accomplished with speech signal capturing, creating a code bank of speech samples, and

then by applying pre-processing techniques.For classifying a wave sample, four layered FFANN model is used with resilient back-propagation (Rprop). The proposed system
yields high accuracy for two and five classes.For db-8 level-5 DWT filter 98.40%, 95.73%, and 95.20% accuracy rate is achieved with 10, 15, and 20 classes, respectively. Haar level-5 DWT filter shows 97.20%, 94.40%, and 91% accuracy ratefor 10, 15, and 20 classes, respectively (Rehmam2015).

In the proposed work, the techniques of wavelet transform (WT) and neural network were introduced for speech based text-independent speaker identification and Arabic vowel recognition. The linear prediction coding coefficients (LPCC) of discrete wavelet transform (DWT) upon level 3 features extraction method was developed. Feature vector fed to probabilistic neural networks (PNN) for classification. The functions of features extraction and classification are performed using the wavelet transform and neural networks (DWTPNN) expert system. The declared results show that the proposed method can make an powerful analysis with average identification rates reached 93. Two published methods were investigated for comparison. The best recognition rate selection obtained was for framed DWT. Discrete wavelet transform was studied to improve the system robustness against the noise of 0dB. Our investigation of speaker-independent Arabic vowels classifier system performance is performed via several experiments depending on vowel type. The declared results show that the proposed method can make an effectual analysis with identification rates may reach 93%(Daqrouq2011).

In this paper, an effective approach for Chinese speech recognition on small vocabulary size is proposed the independent speech recognition of Chinese words based on Hidden Markov Model (HMM). The features of speech words are generated by sub-syllable of Chinese characters. To improve the performance, keyword spotting criterion is applied into the system (Huang2011).

In this paper describes and proposes an efficient and effective framework for the design and development of a speaker-independent continuous automatic Arabic speech recognition system based on a phonetically rich and balanced speech corpus. The speech engine uses 3-emitting state Hidden Markov Models(HMM) for tri-phone based acoustic models. The language model contains both bi-grams and tri-grams(Abushariah2012) .

In this paper, a novel vowel feature extraction method via hybrid wavelet and linear prediction coding (LPC) is presented here. The proposed Arabic vowels recognition system is composed of very promising techniques; wavelet transform (WT) with linear prediction coding (LPC) for feature extraction and feed forward back propagation neural network (FFBPNN) for classification. Moreover, different levels of WT were used in order to enhance the efficiency of the proposed method. Level 2 until level 7 were studied

(Al Azzawi2011).

3. System Overview of Speech Recognition

Speech recognition system generally consists of four main stages are Data acquisition stage, Pre-processing, Feature Extraction ,and Recognition stage.

4. Linear Predication Coding(LPC)

The Linear Predictive Coding presents a compact and precise representation of the spectral magnitude for signals and generates coefficients related to the vocal tract configuration. In LPC, speech sample can be estimated as a linear combination of past samples. Several researches have been performed on Arabic speech recognition using LPC features, Linear predictive coding (LPC) has always been a popular feature due to its accurate estimate of the speech parameters and efficient computational model of speech . One main limitation of LPC features is the linear assumption that fails to take into account of the non-linear effects and sensitivity to acoustic environment and background noise .Since our system will record dataset in calm environment then we will use LPC due to its advantages in non-noisy (silent) system(M. Amer2014).

The Levinson-Durbin algorithm is used to derive the LPC parameter set from an autocorrelation computation with the highest autocorrelation value p, being the order of the LPC analysis(Alaoui2008).

5. Wavelet Transform

The wavelet technique is considered a relatively new technique in the field of signal processing for feature extraction. Wavelet transforms have been used by several researches for automatic speech recognition, speech coding and compression, speech de noising and enhancement and other processes. It replaces the fixed bandwidth of Fourier transform with one proportional to frequency which allows better time resolution at high frequencies than Fourier Transform.

The term wavelet means a small wave. The smallness refers to the condition that this (window) function is of finite length. And as this function is oscillatory so it is called wave.

Wavelet transform was introduced as it is more suitable to deal with non-stationary signals like speech. The advantage of using the Wavelet over the Fourier Transform is that it provides at what time which frequency is present. So it provides time frequency information of the signal. The Fourier Transform (FT) is not suitable for the analysis of such non stationary signal because it provides only the frequency information of signal but does not provide the information about at what time which frequency is present.

So Wavelets have the ability to analyze different parts of a signal at different scales. The wavelet transform (WT) is a transformation that provides time-frequency representation of the signal.

The continuous one dimensional wavelet transform (CWT) is a decomposition of f(t) into a set of basis function a,b(t) called wavelets:

$$w(a,b) = \int f(t)\ \Psi^{*}{}_{a,b}(t)\, dt \qquad (1)$$

The wavelets are generated from a single mother wavelet called $\Psi^{*}{}_{a,b}$ by dilation and translation. As the functions with different region of support that are used in the transformation process are derived from one main function, so it is called the mother wavelet.

$$\Psi_{a,b} = 1/\sqrt{a}\ \Psi((t-b)/a) \qquad (2)$$

Where: f(t) is the signal to be analyzed, a is the scale, and b is the translation factor.Ψ (t) is the transforming function and is called the mother wavelet. Filters of different cut off frequencies are used to analyze the signal. As CWT is a function of two parameters, scale and translation parameters as a=2j and b=2j k. So DWT theory requires two sets of related functions called scaling function and wavelet function given by

$$\phi(t) = \sum_{n=0}^{N-1} h[n]\sqrt{2}\phi(2t-n) \qquad (3)$$

And

$$\Psi(t) = \sum_{n=0}^{N-1} g[n]\sqrt{2}\phi(2t-n) \qquad (4)$$

where, function ψ(t) is called scaling function, h[n] is an impulse response of a low pass filter and g[n] is an impulse response of a high pass filter. The basic operation principles of DWT are similar to the CWT however the difference between them is that the scales used by the wavelet and their positions are sampled up or down by a factor of two. This is called the dyadic scales. The discrete wavelet transform (DWT) uses filter banks for the construction of the multi resolution time-frequency plane.

A filter bank consists of filters which separate a signal into frequency bands. for a two stage filter it consists of a low pass filter L(z) and a high pass one H(z). For many signals, the low-frequency content is the most important part. It is what gives the signal its identity. In wavelet analysis, the high-scale ,low-frequency components of the signal are called the approximations, while the low-scale, high-frequency components are called the details.

The output of the filters each contain half the frequency content, but an equal amount of samples as the input signal. The two outputs together contain the same frequency content as the input signal, however the amount of data is doubled. Therefore down sampling by a factor two, denoted by ↓ 2, is applied to the outputs of the filters in the analysis bank. The process of down sampling is that what produces the DWT coefficients. Given a signal S of length N, the DWT consists of log2 N stages at most. The first step produces, starting from S, two sets of coefficients: approximation coefficients CA1 and the detail coefficients CD1. These vectors are obtained by convolving S with a low pass filter for approximations, and with a high pass filter for details. The decomposition process can be repeated, with successive approximations being decomposed in turn, so that one signal is broken down into many lower resolution components.

There are a lot of wavelet families, the Daubechies are the most widely used in speech recognition problems. The daubechies names are written as dbN where N is the order of the family(Abdalla2013).

6. Preprocessing

Preprocessing is the fundamental signal processing applied before extracting features from speech signal, for the purpose of enhancing the performance of feature extraction. algorithms.

Step one: Preemphasis wave where it is to pass the speech signal on a filter from the type of high- pass filter (HPF) for the purpose of removing the low frequencies of the speech signal .

Step two: Divide incoming signal X(n) to overlapping frames of equal size, each frame consists of (N = 256) of the samples and the amount of overlap (M = 128) samples, from any interference by 50% , the next step after creating frames.

Step three: Applying the window function, on each frame of the speech signal.

7. Endpoint Detection

We use end point detection to extract the speech word and remove the background noise and silence at the beginning and end of the speech word. End point detection improves performance of the system in terms of accuracy and speed. The following steps explain the end point detection algorithm:

Step one: Amplitude rate is calculated for each frame of the speech signal and stored in the matrix .

Step two: Find the highest value and the lowest value of the amplitude matrix rate, which can be represented by Amax, Amin.

Step three: Calculate the threshold as follows:

$$I1 = 0.03 * (Amax - Amin) + Amin \quad (5)$$

$$I2 = 4 * Amin \quad (6)$$

$$Ths = MIN (I1, I2) + 0.4 \quad (7)$$

Step four: Calculate the frames that will be removed from the beginning of the signal depending on the value of the threshold, the deleted frames from values of the amplitude rate represent the frames with amplitude rate less than the threshold, and if we assume that F indicate the number of frames that will be removed from the beginning of the signal , it can calculate the number of samples that are removed as follows:

$$\text{Number of removed frames} =((F-1)*M) + N \quad (8)$$

To remove silence from the end of the word, apply the same previous steps, taking into account the fact that I=1, J= length of matrix A(Sahoo2014).

8. The proposed algorithms for feature extraction in the research

8.1 (LPCS) algorithm

Step one: Apply preprocessing stage on the signal and create the frames, each frame is (256 samples), and this produce a matrix (K*M) so K: number of frames, M: length of frame.

Step two: Find the linear prediction coefficients for each row of the matrix (K*M) and get (20) coefficient which represent features vector for each frame (so the number of columns always have 20 column, while the number of rows which represent number of frames unlimited).

Step three: applying Skewness equation on each column of the matrix , to get a feature vector contain 20 value, so these values represent the input word to recognition techniques. (In order to overcome the resulting problem from the lack of a fixed time to record the words and that was the difference in the number of frames, resulting from recording to another, which leads to a difference in the number of features vectors that can be obtained from each word) as in figure(1).

$$\text{Skewness}=\frac{1}{N}\left(\frac{\sum_{i=1}^{N}(Xi-\bar{X})^3}{\sigma^3}\right) \quad (9)$$

Where as σ^3 represent standard deviation.*3.2*

8.2 (WLPCCS)Algorithm

Step one: Apply preprocessing stage on the signal and create the frames, each frame is (256 samples), and this produce a matrix (K*M) so K: number of frames, M: length of frame.

Step two: Applying wavelet transform on all frames of the speech signal through the passage of each frame on a low frequency filter and high frequency filter depending on Dubitseh filter coefficients (DB). The coefficients that precisely depended after several experiments are (DB8), The filtered signal result are details coefficients (cD) and approximate coefficients (cA), This process continues by continuing segmenting the approximate coefficients cA resulting from each step and to four levels of retail .

Step three: Linear Predictive Coding technique is applied on approximate coefficients cA and details coefficients, so as to get the features vector of each frame, then the vectors stored in the matrix, the number of rows equal number of frames in speech signal, and number of columns equal 20 column(number of extract features).

Step four: Skewness equation is applied to each column in the matrix to get the features vector consists of 20 values as described in the previous algorithm.

Step five: cpestrum technique is applied to the approximation coefficient and detail coefficient for each frame as shown in figure(2).

9. The proposed algorithms for recognition in the research

9.1 Artificial Neural Network

The use of neural network (Classifier) as they provide large-capacity , high-speed retrieval of a large amount of information and outstanding ability in the process of discrimination as a result of building on the style of parallel processing .

9.1.1 Design and Training *Artificial Neural Network*

In this paper, we use a multi-layer feed-forward back propagation network that rely on supervised learning strategy so as to suitability pattern recognition. In the proposed system we suggest a neural network for each proposed method of extraction features, so in order to obtain the best possible results of each method, in the following steps we explain the algorithm used in the training of every network of the proposed networks:

Step one: The data set used in the training phase consist of 900 samples recorded in a clean environment(15 speakers*10 words*6 repetition=900) .

Step two: Extract features and store features vectors in the input matrix(P).

Step three: Configure the target matrix(T).

Step four: Identify the factors influencing in the efficiency of neural network training

Step five: Training feed-forward back propagation network using training pairs(P,T).

Step six: Store ideal weights and network parameters in training file, so as to used it network test.

The experiments showed and numerous attempts relying on the principle of rial and error in the training of all of the network of proposed networks for determining the best factors of the network process , as shown in table (1) . Notes from table(1) two key points, the first point is the presence of two layers for each network of the proposed networks and the reason for this is due to the degree of complexity in the audio data in addition to the input data we have relatively large , matrix capacity (20 * 900) , as observed after experiments that using one hidden layer was not enough to deal with this data , so we used two layers .

The second point is the use of Alsijmaoah function (Logsig) as activated function, this function has been selected after a comparison between different types of activation functions in order to find a function that gives the best results. Has been shown through the application that is better than the rest of the functions are Alsijmaoah activation function .

For the number of input and output nodes to the proposed networks, it was twenty node for input layer which is the number of extract features from each word, and ten nodes for output layer which is the number of recognition words, while the number of nodes in the hidden layers different from one network to another as every network need .

To take the final recognition decision on any network of the proposed networks we have to use a value representing the threshold level, and based on this value the output network is converted which is always between zero and one, because the used function is Alsijmaoah, depending on this zeros and ones a decision is made, and the threshold value which gave the highest decision rate of the proposed networks was (0.4) figure(3) showed the training and recognition stages in neural network .

9.2Fuzzy Neural Network

Hybrid of a neural network with fuzzy logic process takes many forms and different methods including fuzzification data, and then input into the neural network , the use of neural network in modifying membership functions parameters in the fuzzy logic, in addition to the possibility of fuzzification output data of the used neural network this method was used in this research.

The main problem facing us in the decision-making process using a neural network is to determine the value of the threshold, which plays an important role in the efficiency of the process of recognition, since the change of threshold value change ratios for each word, thereby changing the overall recognition rate, for example, you can get a high recognition rate for the first word when a certain threshold used, while required to obtain the highest recognition rate for the second to change the threshold limit value to another value, and so the rest of the words, and this means it is difficult to choose a value representing the threshold limit where the recognition ratios for all words are the best, and to overcome the problem arising from the use of threshold limit, we have been proposed neural network without threshold and replacing it with the concept of fuzzy logic by proposing the use of fuzzy neural network, the following steps illustrate the mechanism that has been followed in the design of the proposed fuzzy neural network:

Step one: Choose the previous neural network that is configured and trained in advance to make an adjustment and the reason for choosing this network being the network that gave the best results.

Step two: Cancel the threshold limit and the conditions that were used in the previous classification process.

Step three: use the output of neural network, whose value is always between zero and one (because of the use of Alsijmaoah function in the output layer) as input to the fuzzy logic.

Step four: fuzzification input data to logic variables (low, mid, high), with a degree between (0-1), by using membership functions shown in figure(4).

Step five: Form 56 rule from (IF-Then) fuzzy rules, depending on the logic variables (low, mid, high), and we choose fuzzy factor (AND) in connection process between multiple conditional elements, we use AND rule in inclusion process, and heavy center rule in the defuzzification process.

10. Results

10.1WLPCC Network Results

In this research in the data collection stage we recorded 10 arabic words with 15 different speakers in a clean environment, each speaker read every word 9 times (6 of them are used in training and the remaining are used in the test phase). Table(2) shows the percentage to recognition each word for all the speakers, and for all the data (training, test).

The percentage rate = (The number of correct recognition words)/(The total number of words) × 100%

To explain the efficiency of the proposed algorithm for feature extraction , we compare it with a classical technique for feature extraction, table(3) explain the result from applying wavelet transform technique and liner predictive technique (WLPCC) on the same speech data base .

It observed during the previous favourable results, using cpestrum analysis, wavelet transform and linear predictive technique together in feature extraction process are better than using cpestrum analysis and linear predictive technique.

10.2FWLPCC Network Results

The following are the test result for the fuzzy neural network(FWLPCC), table(4) explain recognition rate for the training words and non training words figures (5) show a chart for the training data of the networks (WLPCC) and (FWLPCC) while figure (6) show a rate for the non training data of the networks (WLPCC) and (FWLPCC).

11. Conclusions

Fuzzy neural network based speech recognition system is proposed in this research. This system is developed use a mixing between wavelet transform, linear predictive technique, and cpestrum analysis feature extraction methods. In this work effective feature extraction methods for Arabic words is developed, taking in consideration that the computational complexity is very crucial issue.Trying to enhance the recognition process, three techniques were applied for feature extraction stage: LPC, LPCS and WLPCCS, but we choose the wavelet transform technique because gives details are difficult to obtain directly or using traditional methods, it provides information on the wave in all of the time and frequency. So using filter Daubachies 8 is the most suitable to deal with the speech wave after compered with other filters. Two techniques were applied for recognition stage: neural network and fuzzy neural network. The experimental results on a subset database showed that feature extraction method and recognition method in this research appropriate for Arabic recognition system. Our investigation of Arabic words recognition system performance is performed via several experiments depending on Arabic type. The declared results showed that the proposed method can make an effectual analysis with identification rates may reach 100% in some cases.

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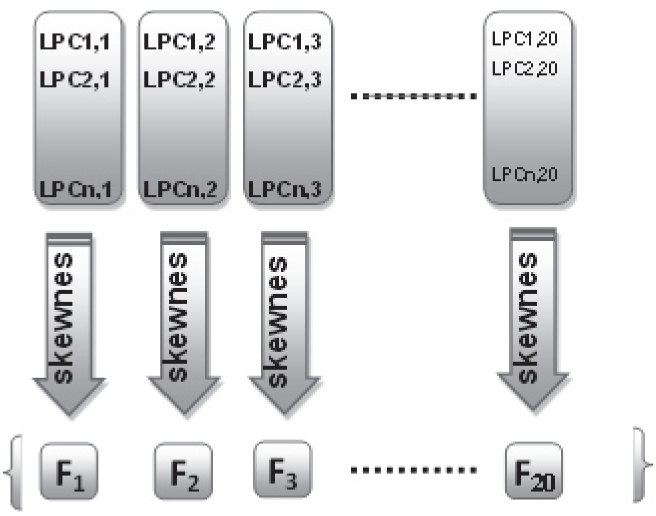

Figure 1. Applying Skewnes equation

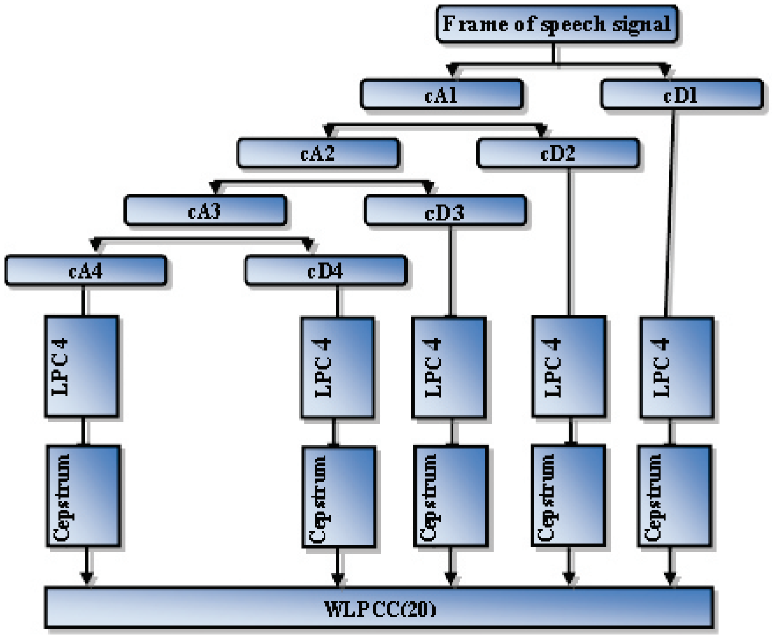

Figure 2. Applying WLPCC Algorithm on one frame

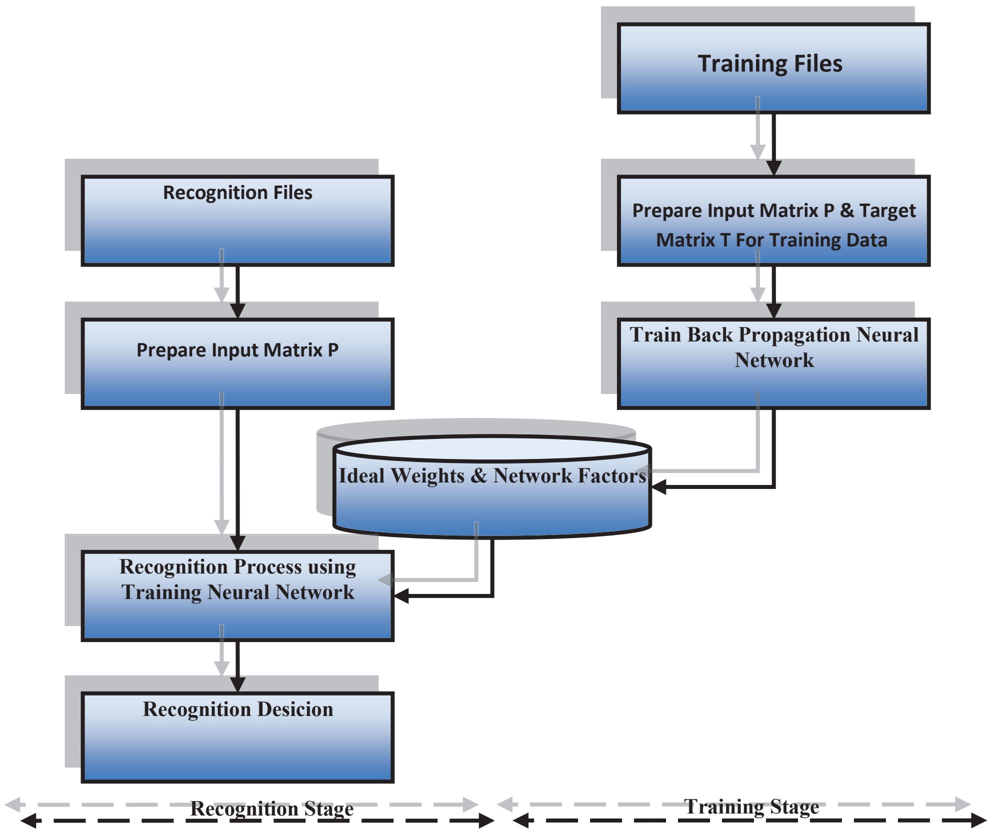

Figure3: Diagram Explaining Training & Recognition Stages in Neural Network

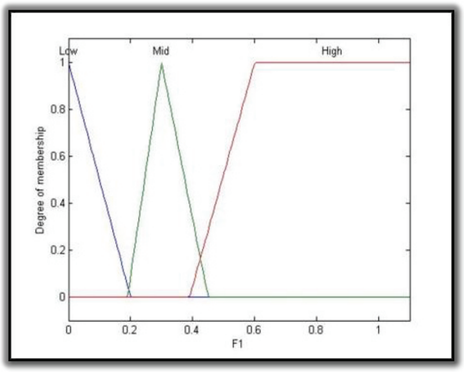

Figure4. membership functions used in fuzzification process

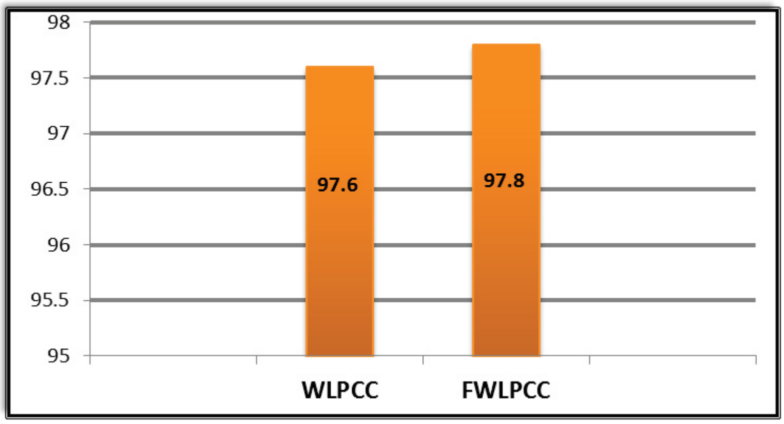

Figure5: chart for the training data of the networks (WLPCC) and (FWLPCC)

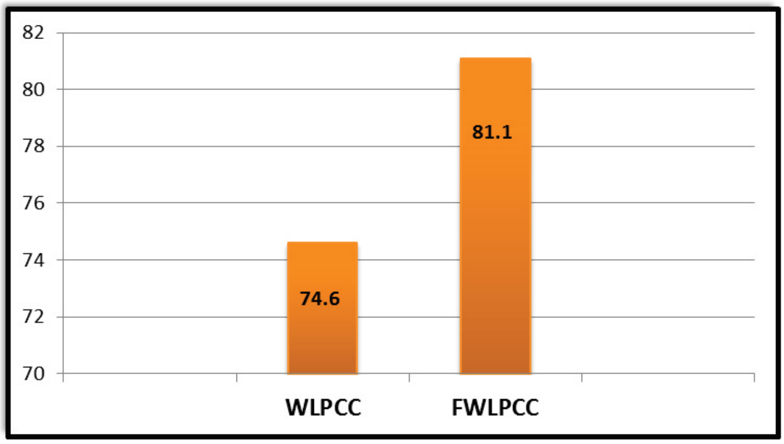

Figure6: chart for the non training data of the networks (WLPCC) and (FWLPCC)

Table(1) Best factors in each network

WLPCC network	LPC network	factor /network
50	70	Number of nodes in the first hidden layer
50	80	Number of nodes in the second hidden layer
Logsig	Logsig	Activation function for the first hidden layer
Logsig	Logsig	Activation function for the second hidden layer
Logsig	Logsig	Activation function for the output layer
10000	250000	Total number of courses
58 minute	2.4 hour	Total time
0.000182	0.00402	Square error rate

Table(2) Recognition percentage for each word using the proposed algorithm (LPC)

Recognition percentage for training words											
Total percentage	أحزان	يمين	يسار	لندن	الشارقة	يتكلم	ياسين	خديجة	محمد	الله	word
96.5%	98.9	100	97.8	90	100	96.7	90	95.6	98.9	97.8	Percentage
Recognition percentage for non training words											
Total percentage	أحزان	يمين	يسار	لندن	الشارقة	يتكلم	ياسين	خديجة	محمد	الله	Word
54.2%	57.8	66.7	57.8	31.1	71.1	44.4	73.3	68.9	37.8	33.3	Percentage

Table(3) Recognition percentage for each word using the proposed algorithm (WLPCC)

Recognition percentage for training words											
Total percentage	أحزان	يمين	يسار	لندن	الشارقة	يتكلم	ياسين	خديجة	محمد	الله	word
%97.6	100	100	100	95.6	98.9	97.8	87.8	100	97.8	98.9	Percentage
Recognition percentage for non training words											
التمييز نسبة الكلية	أحزان	يمين	يسار	لندن	الشارقة	يتكلم	ياسين	خديجة	محمد	الله	Word
74.6%	80.0	88.9	66.7	60.0	77.8	66.7	62.2	77.8	88.9	77.8	Percentage

Table(4) Recognition percentage ratios for each word using fuzzy neural network

Recognition Percentage ratio for training words											
Total percentage	أحزان	يمين	يسار	لندن	الشارقة	يتكلم	ياسين	خديجة	محمد	الله	الكلمة
97.8 %	**100**	**100**	**100**	**95.6**	**98.9**	**100**	**87.8**	**100**	**97.8**	**98.9**	**percentage**
Recognition Percentage ratio for non training words											
Total percentage	أحزان	يمين	يسار	لندن	الشارقة	يتكلم	ياسين	خديجة	محمد	الله	الكلمة
81.1 %	**80.0**	**91.1**	**75.6**	**66.7**	**80**	**73.3**	**71.1**	**88.9**	**97.8**	**86.7**	**Percentage**