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Comparative Analysis of Methods Used to Extract Speech Signal Features

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Abstract: The stage of extracting the features of the speech file is one of the most important stages of building a system for identifying a person through the use of his voice. Accordingly, the choice of the method of extracting speech features is an important process because of its subsequent negative or positive effects on the speech recognition system. In this paper research we will analyze the most popular methods of speech signal features extraction: LPC, Kmeans clustering, WPT decomposition and MLBP methods. These methods will be implemented and tested using various speech files. The amplitude and sampling frequency will be changed to see the affects of changing on the extracted features. Depending on the results of analysis some recommendations will be given.
Keywords: Speech signal, Amplitude, FS, LPC, Kmeans, WPT, MLBP, features**.**

1. Introduction

The digital speech file is considered one of the important digital data due to its use in many vital applications, including protection and security systems [18-20], as this application requires the identification of the human being through a word or a speech sentence generated by this human [1-6]. A speech file consists of a set of speech samples taken in a specific period, each sample has a value representing the speech amplitude (A) in a specific period of time, the sample values are usually organized in one column matrix (mono speech), or two column matrix (stereo speech). The speech file size depends on the recording time and the sampling frequency (FS), this size will be increased if the recording time and/or FS increased, and the parameters A and FS can be easily changed or may be changed depending on the way of pronouncing the speech file by the person, as shown in figure 1[30-35].

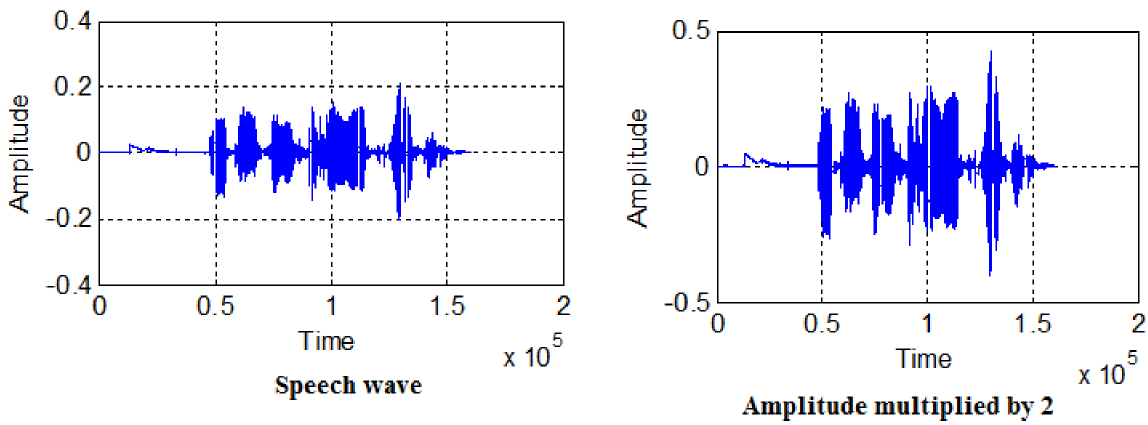

Figure 1: Same speech signals with various amplitudes

The size of the speech file is usually very large, which requires effort and time to identify it through speech recognition and discrimination systems [7-11], which will negatively affect the efficiency of the recognition system. Therefore, it is necessary to search for a way to represent the speech file with a small set of values called speech features as shown in figure 2, and using these features, it is possible to easily and in a very short time identify the speech file [36-40].

The process of creating the features of the different speech files is the first and important stage for building a system to identify the person through his voice, where the features are extracted for each speech file and these features are preserved in the features database, which will be used later to train the recognition system as shown in figure 3.

The extracted features for any used method must satisfy the following requirements [12-17]:

1) The number of values in the features array must be small.
2) The number of values must be fixed.
3) The set of values for each person-spoken phrase must be unique, this make the recognition system capable to identify the person and the spoken phrase.
4) The values must be numeric to facilitate easy processing.
5) The features must be stable even if the amplitude or sampling frequency was changed as results of multiplying or dividing them by a constant as shown in figure 4.

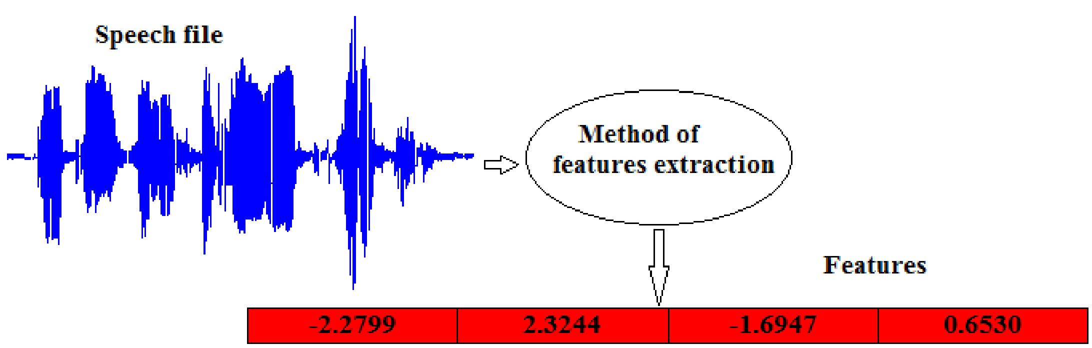

Figure 2: Features to represent speech file

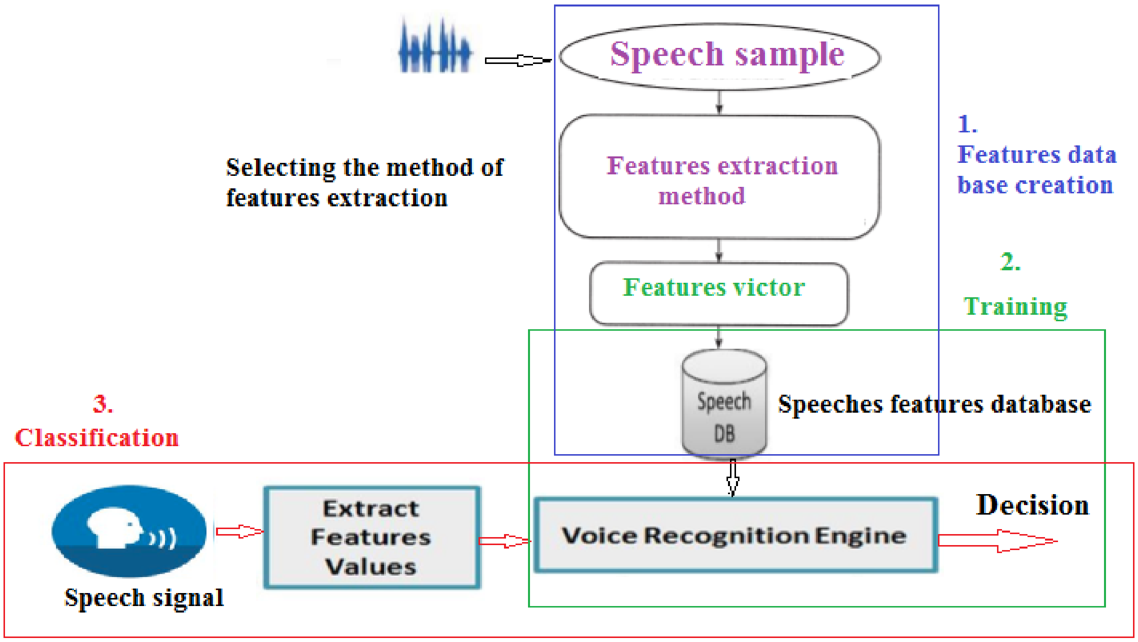

Figure 3: Speech recognition process

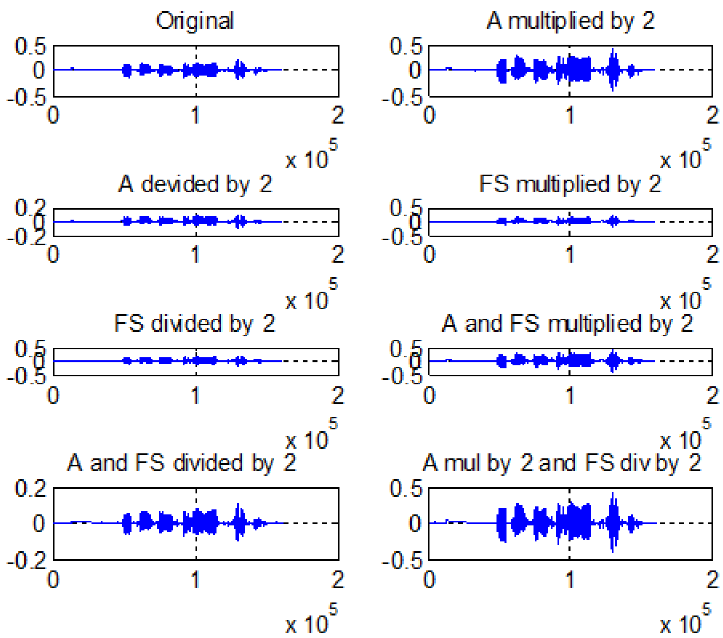

Figure 4: Speech signal example with different A and FS

2. Speech features extraction methods

- Using LPC coefficients as features

Any current speech sample can be represented by the past values using LPC coefficients [43], these coefficients relate the pass p samples of the speech x to the current value using equation 1:

$$\hat{x}(n) = -a(2)x(n-1) - a(3)x(n-2) - \cdots - a(p+1)x(n-p) \quad \textbf{(1)}$$

LPC coefficients can be easily obtained applying the matlab function lpc and setting the number of coefficients to 4, these coefficients can be easily used as features for the selected input speech signal, as shown in example shown in figure 5:

```
x=rand(1,4096);
a=lpc(x,4)

a =

            a(2)       a(3)       a(4)       a(5)
  1.0000  -0.2238    -0.2197    -0.2592    -0.2192
```

Features

Figure 5: LPC coefficients as features

- Using Kmeans clustering

Kmeans clustering means dividing the input data set into groups called clusters, the centroids of the clusters here can be used as the data features as shown in figure 6.

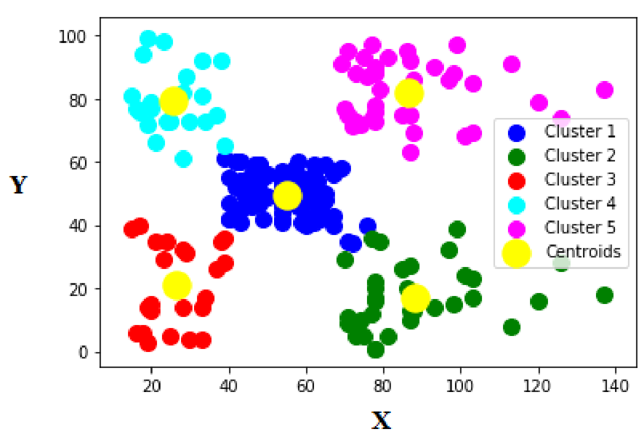

Figure 6: Data clustering

Kmeans clustering is a simple way of speech signal features extraction [23], [24]. To get a 4 elements features victor we have to initialize the number of clusters to 4 and give each cluster center an initial value, the reshaped to one row speech signal must be used as an input data set. The clustering process will be performed in many passes, each pass apply the following sequence of operations:

- For each sample find the distance to each centroid (the absolute value of the sample value subtracted from the centroid).

- The minimal distance means that the sample belongs to the associated cluster.
- Find the averages samples of each cluster.
- Update the centroids to the averages samples.
- If there is any change in the centroids values repeat another pass.

Table 1 shows an example of data items clustering, using 2 clusters, and initial centroids (2 and 3):

Table 1: Data clustering example

Pass 1					Pass 2			
Data item	D1	D2	Cluster	New centroids	D1	D2	Cluster	New centroids
3	1	0	**2**	C1=2; C2=8	1	5	**1**	C1=2.5; C2=9.33
12	10	9	**2**		10	4	**2**	
7	5	4	**2**		5	1	**2**	
10	8	7	**2**		8	2	**2**	
2	0	1	**1**		0	6	**1**	
Pass 3								
3	0.5	6.33	**1**	C1=2.5; C2=9.33 No change: stop				
12	8.5	2.67	**2**					
7	4.5	2.33	**2**					
10	7.5	0.67	**2**					
2	0.5	7.33	**1**					

- **Using WPT decomposition**

Wavelet packet tree [25-27] uses the concepts of binary tree to decompose the speech signal. Here the speech signal will be used as a tree root (see figure 7), using Haar equations (equation 2 and 3) the signal will be decomposed into approximation and detail, each of them will have a size equal to the root size divided by 2 [41], [42].

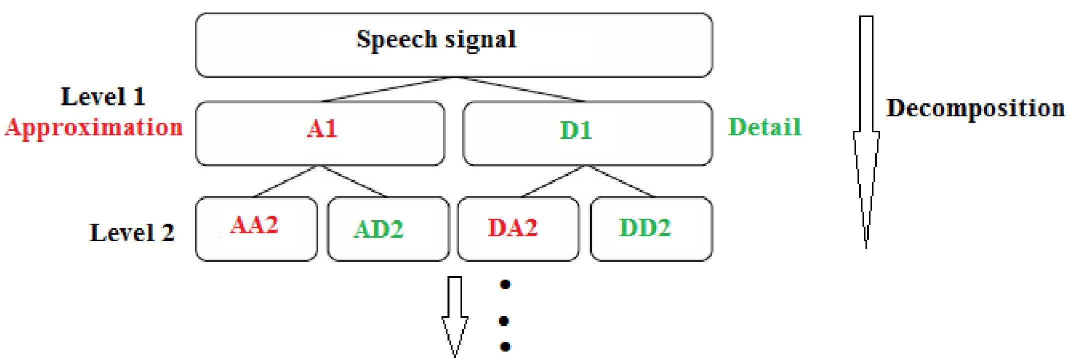

Figure 7: Speech signal decomposition

Haar scaling (low pass) function:

$$s_{j+1,i} = \frac{even_{j,i} + odd_{j,i}}{\sqrt{2}} \quad \text{Approximation} \qquad (2)$$

Haar wavelet (high pass) function:

$$d_{j+1,i} = \frac{even_{j,i} - odd_{j,i}}{\sqrt{2}} \quad \text{details} \qquad (3)$$

Figures 8 and 9 show an example of signal decomposition

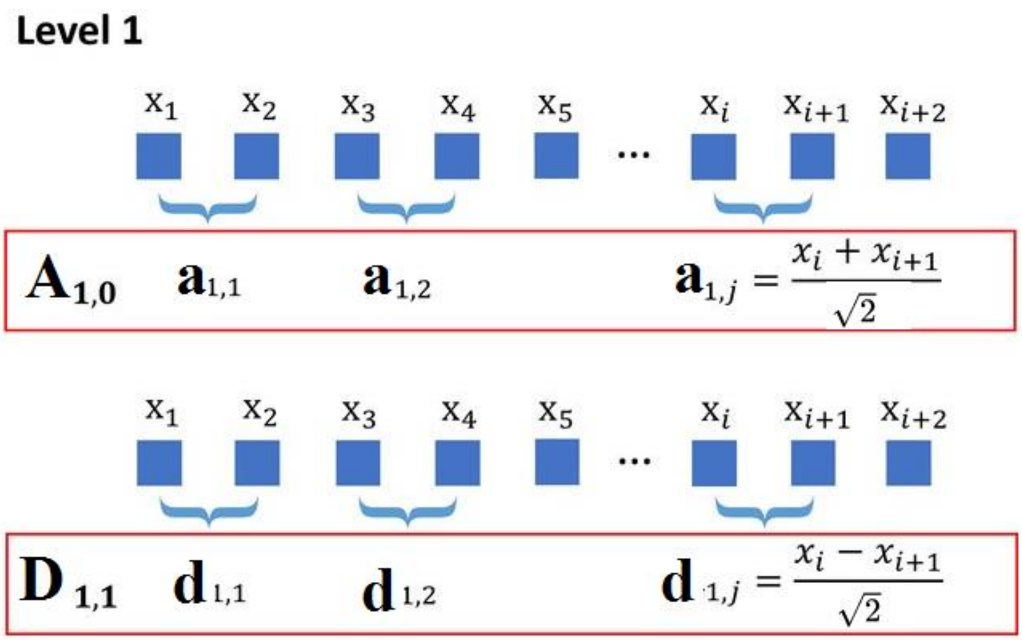

Figure 8: Approximation and detail calculation

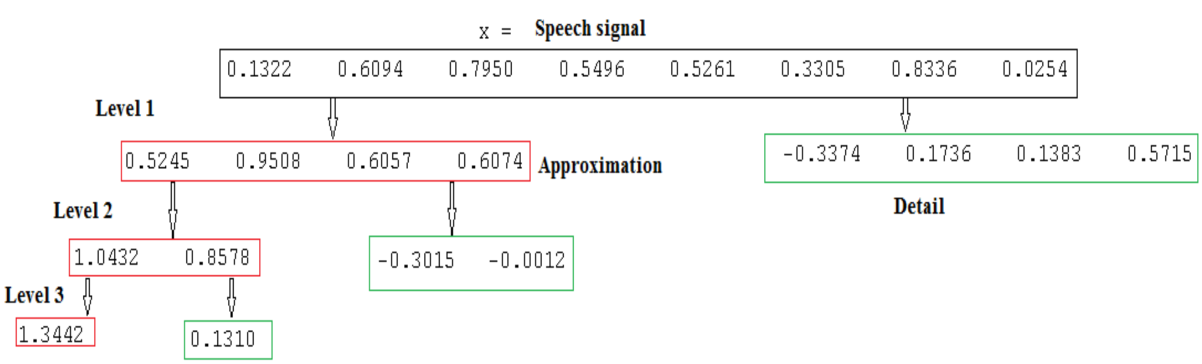

Figure 9: 8 samples decomposition

The main disadvantage of this method is selecting the number of levels. The speech signals usually have various sizes and they may be not of multiple 2, to avoid this problem we can take a part of the speech signal with multiple of 2 size (let us say with size =2048 samples), this part can be decomposed to get an approximation of 4 elements.

- **Using local binary pattern operator**

Local binary pattern (LBP) method was adapted to digital images to calculate LBP image [21], [22], each pixel in the input image will be treated as shown in figure 10 to get the LBP image.

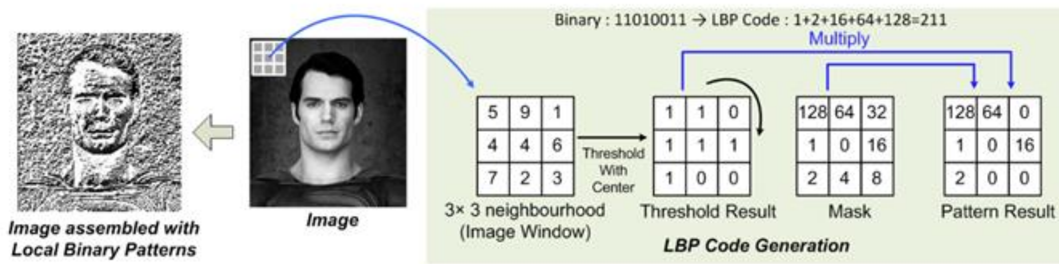

Figure 10: LBP image calculation

Based on the LBP method we can use the method explained in figure 11 to create a 4 elements features victor for a speech signal, each element will indicate the total repetition of the associated value (0, 1, 2, and 3), we will refer to this method as MLBP.

			P(i,i-1)	P(1,i)	P(1,i+1)				
....	120	200	150	177	213	134	167	100	.....

P(1,i-1)>=P(1,i)=0 so a0=0

P(1,i+1)>=P(1,i)=1 so a1=1

Binary=10 decimal=2,

so add 1 to the repetition of 2

Figure 11: Creating 4 elements features victor

3. Methods implementation

4 spoken phrases for 4 persons (shown in table 2) were recorded and the obtained speech signals were used in various experiments.

Table 2: Spoken phrases

Spoken phrase	**Speech contents**
1(S11, S21,S31,S41)	I can finish the home work very fast
2(S12, S22,S32,S42)	Good morning
3(S13, S23,S33,S43)	How are you
4(S14, S24,S34,S44)	I am a computer engineer

The first person speech signals processed using the above mentioned methods; table 3 shows the obtained experimental results:

Table 3: Extracted features for person 1 speeches

Speech		Features			
S11	LPC	-2.2799	2.3244	-1.6947	0.6530
S12		-2.1706	2.0174	-1.4781	0.6337
S13		-2.3294	2.2694	-1.4956	0.5576
S14		-2.3351	2.3032	-1.5264	0.5605
S11	Kmeans	-0.0752	-0.0287	0.0031	0.0719
S12		-0.0448	0.0004	0.0247	0.0935
S13		-0.0591	0.0001	0.0292	0.1082
S14		-0.0642	-0.0001	0.0475	0.1155
S11	WPD	0.0736	0.0013	-0.0106	0.0016
S12		0.1059	0.0193	-0.0027	0.0011
S13		0.1093	-0.0161	-0.0059	-0.0013
S14		0.0511	-0.0035	0.0189	0.0064
S11	MLBP	20146	55914	48912	35794
S12		12207	31151	28426	28566
S13		15284	33729	31175	33474
S14		30434	71067	64660	48877

The first spoken phrase for the 4 persons then were taken and processed using each of the features extraction method, table 4 shows the obtained experimental results.

Table 4**:** Extracted features for spoken phrase 1 for each person

Speech		Features			
S11	LPC	-2.2799	2.3244	-1.6947	0.6530
S21		-2.3615	2.4737	-1.6856	0.5764
S31		-2.2712	2.4174	-1.8536	0.7146
S41		-1.2251	0.8171	-0.7701	0.4952
S11	Kmeans	-0.0752	-0.0287	0.0031	0.0719
S21		-0.0381	-0.0009	0.0241	0.0584
S31		-0.0290	-0.0009	0.0197	0.0722
S41		-0.2935	-0.0655	0.0843	0.2931
S11	WPD	0.0736	0.0013	-0.0106	0.0016
S21		-0.0034	-0.0092	-0.0063	-0.0041
S31		0.0007	0.0051	-0.0030	-0.0004
S41		0.0908	0.0262	-0.2476	0.0732
S11	MLBP	20146	55914	48912	35794
S21		7505	22671	25398	30440
S31		9376	30476	30419	12607
S41		7323	15801	14305	9884

Spoken phrase 1 for person 1 then was taken, the amplitude and sampling frequency was changed various times, the updated speech signal were taken and treated using each of the 4 methods, the obtained results are shown in tables 5 thru 8:

Table 5: Updated speech 1 of person 1 features using LPC method

Speech	Action	Features			
S11	Original	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	Amplitude multiplied by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	Amplitude divide by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	FS multiplied by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	FS divided by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	Amplitude and FS multiplied by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	Amplitude and FS divided by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**
S11	Amplitude multiplied by 2 and FS divided by 2	**-2.2799**	**2.3244**	**-1.6947**	**0.6530**

Table 6: Updated speech 1 of person 1 features using Kmeans clustering method

Speech	Action	Features			
S11	Original	-0.0752	-0.0287	0.0031	0.0719
S11	Amplitude multiplied by 2	-0.1016	-0.0025	0.0495	0.1651
S11	Amplitude divide by 2	-0.0254	-0.0006	0.0123	0.0412
S11	FS multiplied by 2	-0.0508	-0.0012	0.0248	0.0826
S11	FS divided by 2	-0.0754	-0.0288	0.0031	0.0718
S11	Amplitude and FS multiplied by 2	-0.1016	-0.0025	0.0495	0.1651
S11	Amplitude and FS divided by 2	-0.0376	-0.0143	0.0016	0.0359
S11	Amplitude multiplied by 2 and FS divided by 2	-0.1508	-0.0576	0.0062	0.1437

Table 7: Updated speech 1 of person 1 features using WPT method

Speech	Action	Features			
S11	Original	**0.0736**	**0.0013**	**-0.0106**	**0.0016**
S11	Amplitude multiplied by 2	**0.0368**	**0.0006**	**-0.0053**	**0.0008**
S11	Amplitude divide by 2	**0.0736**	**0.0013**	**-0.0106**	**0.0016**
S11	FS multiplied by 2	**0.0736**	**0.0013**	**-0.0106**	**0.0016**
S11	FS divided by 2	**0.1472**	**0.0027**	**-0.0212**	**0.0032**
S11	Amplitude and FS multiplied by 2	**0.1472**	**0.0027**	**-0.0212**	**0.0032**
S11	Amplitude and FS divided by 2	**0.0368**	**0.0006**	**-0.0053**	**0.0008**
S11	Amplitude multiplied by 2 and FS divided by 2	**0.1472**	**0.0027**	**-0.0212**	**0.0032**

Table 8: Updated speech 1 of person 1 features using MLBP method

Speech	Action	Features			
S11	Original	**20146**	**55914**	**48912**	**35794**
S11	Amplitude multiplied by 2	**20146**	**55914**	**48912**	**35794**
S11	Amplitude divide by 2	**18554**	**55441**	**48430**	**38341**
S11	FS multiplied by 2	**20146**	**55914**	**48912**	**35794**
S11	FS divided by 2	**20146**	**55914**	**48912**	**35794**
S11	Amplitude and FS multiplied by 2	**20146**	**55914**	**48912**	**35794**
S11	Amplitude and FS divided by 2	**18554**	**55441**	**48430**	**38341**
S11	Amplitude multiplied by 2 and FS divided by 2	**20146**	**55914**	**48912**	**35794**

4. Methods analysis

From the obtained experimental results we can raise the following important points:

- ✓ The 4 methods can be used to create a features capable to identify the spoken phrase by any person (see table 3), each spoken phrase has a unique features victor.
- ✓ The 4 methods can be used to create a features capable to identify the person (see table 4), each person speech has a unique features victor.
- ✓ The 4 methods can be used to create a features capable to identify the spoken phrase-person by any person-speech (see table 3, and 4), each spoken phrase-person has a unique features victor.
- ✓ If the amplitude and/or the sampling frequency were changed by multiplying and/or dividing each of them by a constant the best method to be used here is LPC method of features extraction, here the features will be stable and changing the speech signal amplitude and/or the sampling frequency will not affect the extracted features(see table 5).
- ✓ Updating the speech file will affect the extracted features using Kmeans, WPT, and MLBP methods, thus the updated speeches will be treated as a new speech files, these files must be saved alone, and each of these files must treated as a separate file with a separate features. This will adds an extra efforts requiring more memory space to store the speech files and extra entries in the features database (see tables 6, 7, and 8).

For efficiency analysis the average extraction time and throughput (number of samples treated in second) were calculated for the four methods, table 9 shows the obtained results:

Table 9: Efficiency parameters

Method	Average extraction time(seconds)	Throughout(samples per second)
LPC	0.133000	3.5575e+005
Kmeans	0.585000	8.0880e+004
WPT(samples=2048)	0.489000	4.1881e+003
MLBP	0.002000	2.3657500e+007

From table 9 we can see that the efficiency parameters of LPC method are acceptable even if it comes after MLBP method, because MLBP method require a new features for a new updated speech signal.

Conclusion

Several methods of speech signal features extraction methods were experimented, the obtained results were analyzed. It was shown that the best method to be used for features extraction is LPC method. This method is efficient by providing small extraction time and high throughput. LPC can be easily used to identify the person depending on the spoken speech, and identify the spoken-word or spoken phrase by s person. LPC method treats the updated speech files as one file keeping the extracted features stable.

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