Available online at http://jurnal.ahmar.id/index.php/asci

Journal of Applied Science, Engineering, Technology, and Education

ISSN 2685-0591 (Online)

Journal of Applied Science, Engineering, Technology, and Education Vol. 2 No. 1 (2020)
https://doi.org/10.35877/454RI.asci2151

# Evaluation of speech signal features extraction methods

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**Abstract**

Human speech digital signals are famous and important digital types, they are used in many vital applications which require a high speed processing, so creating a speech signal features is a needed issue. In this research paper we will study more widely used methods of features extraction, we will implement them, and the obtained experimental results will be compared, efficiency parameters such as extraction time and throughput will be obtained and a speedup of each method will be calculated. Speech signal histogram will be used to improve some methods efficiency.



*Keywords:* Speech, features, histogram, LBPM, LPCM, WPDM, KMC, speedup, throughput.

## 1. Introduction*

The speech signal, as it emerges from a speaker's mouth, nose and cheeks, is a one-dimensional function (air pressure) of time. Microphones convert the fluctuating air pressure into electrical signals, voltages or currents, in which form we usually deal with speech signals in speech processing, speech signal is emerges from a speaker’s mouth, nose and cheeks, is a one-dimensional function (air pressure) of time [1], [2],[20]. Microphones convert the fluctuating air pressure into electrical signals, voltages or currents, in which form we usually deal with speech signals in speech processing [17], [18], [19]. Human speech is an analogue signal which can be converted to digital signal by applying sampling and quantization as shown in figure 1.

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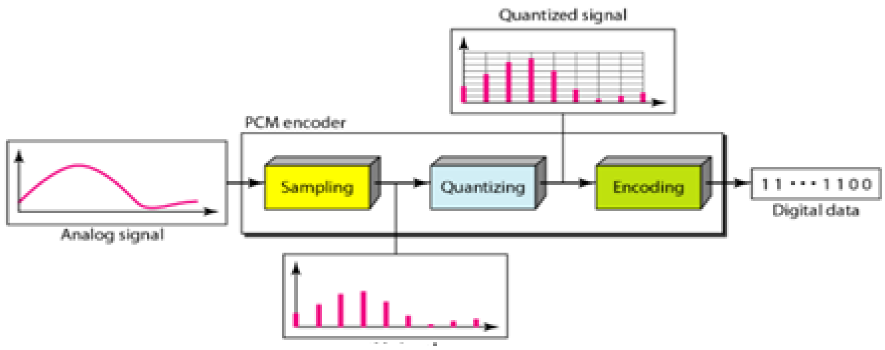


Figure 1: Converting speech analogue signal to digital

Speech signal is an important digital data type due to the vital applications requiring this kind of data, these applications such as security systems application [3], [4] require a high speed of implementation, but the speech signals usually have a big size, and thus will negatively affects the system efficiency and here we will seek a method to represent the speech by a small number of values to increase the process of speech manipulation. Speech signal file size depends on the recording time and the sampling rate[7], [8]. The sampling frequency or sampling rate, $f_s$, is the average number of samples obtained in one second (samples per second), thus $f_s = 1/T$. Table 1 shows some information about the speech signals which we will investigate in this paper [7], [8], [9].

Table 1: Used speech signal files

| Speech # | Spoken words | Fs | Time(seconds) | Size(samples) | Size(bytes) |
|---|---|---|---|---|---|
| 1 | Aqaba is a beautiful city, it is located on the red sea | 44100 | 5.7832 | 255037 | 2040296 |
| 2 | Stay home stay safe | 44100 | 2.8451 | 125469 | 1003752 |
| 3 | Albalqa applied university | 44100 | 3.5109 | 154829 | 1238632 |
| 4 | Amman is the capital city of Jordan | 44100 | 4.1620 | 183544 | 1468352 |
| 5 | How are you | 44100 | 1.9204 | 84691 | 677528 |
| 6 | My name is Ziad | 44100 | 2.5021 | 110344 | 882752 |
| 7 | Please open the door | 44100 | 2.5362 | 111848 | 894784 |
| 8 | Please shut down the computer | 44100 | 3.3558 | 147990 | 1183920 |
| 9 | Speech signal analysis | 44100 | 2.9507 | 130127 | 1041016 |
| 10 | Good by | 44100 | 1.6909 | 74569 | 596552 |
| | Average | | 3.1257 | 137840 | 1102800 |

From table 1 we can see that the average number of samples is big, so the average file size is also big, and this will lead to extra time to identify the speech, so we can represent the speech file by a histogram [12], [13],[14] of 256 values and with size equal 2048 bytes for each speech file[5], [6], [9].

The speech file histogram can be calculated based on local binary pattern (LBP) operator calculation [24], [25], and here we introduce the following method as shown in table 2 to calculate LBP histogram for each speech file.

Table 2: LBP histogram calculation

| Speech samples | X(i-4) | X(i-3) | X(i-2) | X(i-1) | X(i) | X(i+1) | X(i+2) | X(i+3) | X(i+4) | …. | …. |
|---|---|---|---|---|---|---|---|---|---|---|---|
| Values | -0.2 | 1 | 0 | 0.5 | 0.1 | -1 | 0.25 | -.015 | -1 | | |
| Weights | 1 | 2 | 4 | 8 | - | 16 | 32 | 64 | 128 | | |
| Binary | 0 | 1 | 0 | 1 | - | 0 | 1 | 0 | 0 | | |
| Binary = 00101010 decimal =42<br>So add 1 to the index 42(repetition of 42) | | | | | | | | | | | |

Figure 2 shows speech signal 1 and the associated LBP histogram.

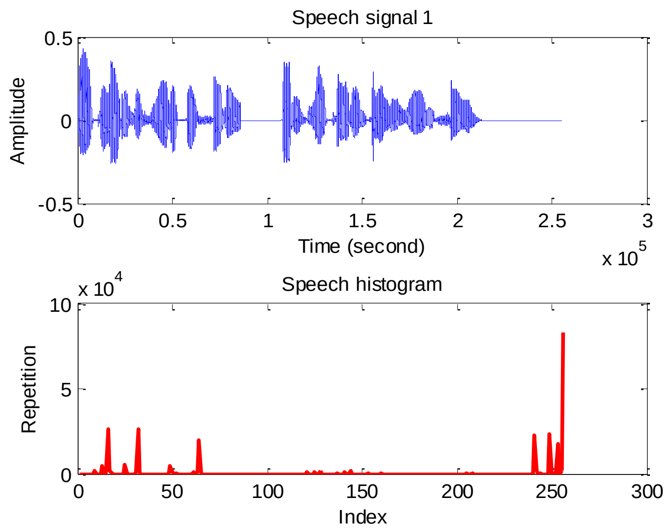


Figure 2: Speech 1 and LBP histogram

To reduce the number of values used to represent the speech signal file we have to seek a method to extract a set of features values [17], [18],[19,[20], which must be unique and small and easily used to identify the speech file.

## 2. Evaluation of methods used for features extraction

### *2.1 LBP based method*

One of the most papular features extraction methods is LBP based method[21], [22],[23], [24], the features will be extracted based on LBP operator and as shown in table 3, by this modified LBP method we can generate a 4 elements features array [11[, [15], [16]for each speech file.

Table 3 Calculating modified LBP method (MLBP) features

| Sample | X(i-2) | X(i-1) | X(i) | X(i+2) | X(i+2) | …….. |
|---|---|---|---|---|---|---|
| Value | 0.95 | -0.75 | - | 0.45 | 0.9 | |
| | <= | <= | | | | |
| Binary | 0 | 1 | | | | |
| Weight | 1 | 2 | | | | |
| | Binary = 10 decimal = 2 so add 1 to features with index = 2 | | | | | |

### *2.2 Wavelet packet decomposition (WPD) for features extraction*

This method of features extraction is based on wavelet packet tree (WPT) [33], [34], [35]to decompose a digital signal into approximations and details as shown in figure 3, here in our paper we will take the approximation of each level of decomposition, the approximation and details packets can be calculated using the following formulas:

$$\mathbf{A}_{j+1,i} = \frac{even_{j,i} + odd_{j,i}}{2} \qquad \mathbf{1}$$

$$\mathbf{D}_{j+1,i} = \frac{even_{j,i} - odd_{j,i}}{2} \qquad \mathbf{2}$$

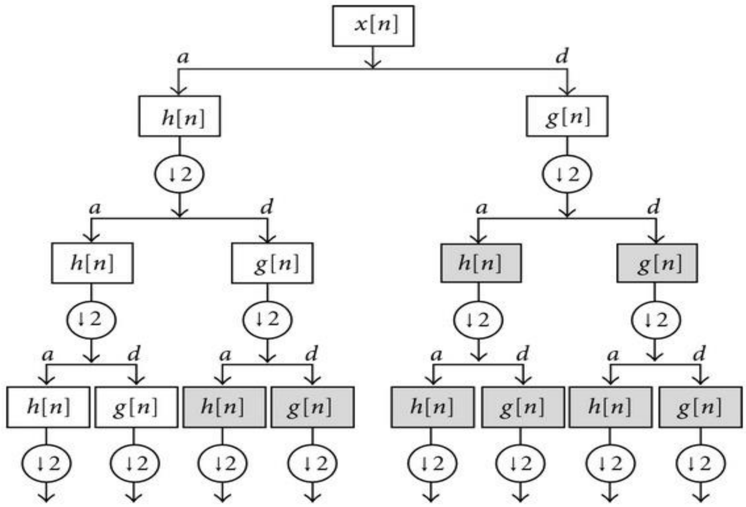


Figure 3: Speech signal decomposition using WPT

Table 4 shows an example of how to calculate the approximation packets at each level.

Table 4: Approximation packets calculations example

| | **2** | **4** | **-2** | **6** | **10** | **12** | **8** | **7** | **-4** | **9** | **3** | **12** | **8** | **0** | **2** | **4** |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Level 1 | Approximation A10 | | | | | | | | Detail D10 | | | | | | | |
| | **3** | **2** | **11** | **7.5** | **2.5** | **7.5** | **4** | **3** | **-1** | **-4** | **-1** | **0.5** | **-6.5** | **-4.5** | **4** | **-1** |
| Level 2 | A20 | | | | D20 | | | | A21 | | | | D21 | | | |
| | **2.5** | **9.25** | **5** | **3.5** | **-0.5** | **1.75** | **-2.5** | **0.5** | **-2.5** | **-0.25** | **-5.5** | **1.5** | **1.5** | **-0.75** | **-1** | **2.5** |
| Level 3 | A30 | | D30 | | A31 | | D31 | | A32 | | D32 | | A33 | | D33 | |
| | **5.875** | **4.25** | **-3.375** | **0.75** | **0.625** | **-1** | **-1.125** | **-1.5** | **-1.375** | **-2** | **-1.125** | **-3.5** | **0.375** | **0.75** | **1.125** | **-1.75** |
| Level 4 | **A40** | **D40** | **A41** | **D41** | **A42** | **D42** | **A43** | **D43** | **A44** | **D44** | **A45** | **D45** | **A46** | **D46** | **A47** | **D47** |
| | **5.0625** | **0.8125** | **-1.8125** | **-2.0625** | **-0.1875** | **0.8125** | **-1.3125** | **0.1875** | **-1.6875** | **0.3125** | -2.3125 | 1.1875 | 0.5625 | -0.1875 | -0.3125 | 1.4375 |

Here we have to notices that to get a 4 element features array for each speech file the number of used levels will be deferent from one file to another because the file size is not fixed. To fix the number of levels (to 6) we can use the speech file histogram, this will be illustrated in the implementation part, and we will refer to this method as WPDH.

### *2.3 K-mean clustering for features extraction*

K-mean clustering (KMC) method [29],[30],[31],[32]divides the speech signal into groups (clusters), each cluster has a centroid , a set of values and we can use the clusters centroids, or the number of values within each cluster, or the within clusters sums as a features. The number of features in features vector depends on the selected number of clusters. Here we can also use speech histogram as an input data set to the clustering process and we will refer to this method later as KMCH.

### *2.4 Using FIR filter to create speech signal features*

Finite impulse response (FIR) filter [25],[26] is a filter with no feedback in its equation. This can be an advantage because it makes an FIR filter inherently stable (see figure 4). Another advantage of FIR filters is the fact that they can produce linear phases. If an application requires linear phases, the decision is simple; an FIR filter must be used.

$$y[n] = b_0x[n] + b_1x[n-1] + b_2x[n-2] + b_3x[n-3]$$

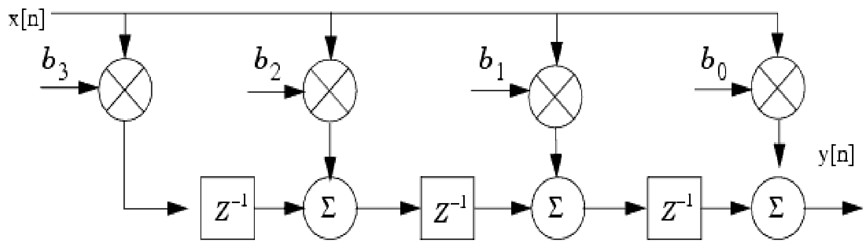


Figure 4. 3 order FIR filter

FIR filters can be used for low-pass filtering, high-pass filtering, band-pass filtering, band-stop (notch) filtering, and other designs. It can be used also to create features for any digital signal such as human speech signal.

The filter coefficients can be extracted using linear prediction coding, these coefficients can be used as a signal features, they are also can be used to reconstruct the signal again. Matlab provides a special function capable to produce FIR filter coefficients, the number of coefficients will equal the selected filter order, and it can be varied from 1 to any defined number, giving us the flexibility to define the size of the created features array. In our paper we will fix the filter order to 4 to get a 4 elements features vector.

### *2.4 Hydraulic Modeling Using Hec-Ras*

Hec-Ras (Hydrologic Engineering System River Analysis System) is used to determine the phenomenon of hydraulic behavior of flow in the channel / river and long-storage object of study by means of simulation / numerical analysis that is able to describe the condition of existing rivers and plans. The scope of Hec-Ras is to calculate water level profiles by modeling steady and unsteady flow, and calculation of sediment transport. The most important element in Hec-Ras is the availability of transverse or longitudinal river geometry.

This software makes it easy for users with a graphical display. In general, Hec-Ras provides the following functions:
• File management
• Data input and editing
• Hydraulic analysis
• Outputs (tables, graphs, figures)

In this study the analysis was carried out using steady flow. Analysis was carried out to determine the profile of the water level and the ability of the river to flow through the discharge. The modeling steps are as follows:
1. Make a schematic of a river network that will be modeled based on the results of field measurements.
2. Entering river geometry data.
3. Define boundary conditions that will be used in the analysis.
4. Enter the flood discharge plan
5. Running a modeling program.
6. Print the results / output.

## 3. Implementation and experimental results

### *3.1 LBPM implementation*

This method was implemented using the speech files shown in table 1, the features for each file were extracted, the extraction time was obtained and the throughput of the method was calculated( throughput is the number of samples processed per second), table 5 shows the results of implementation.

Table 5 LBPM method results

| Speech # | Features | | | | Extraction time(sec) | Throughput(samples per second) |
|---|---|---|---|---|---|---|
| 1 | 70274 | 29265 | 1998 | 153496 | 0.0170 | 15002000 |
| 2 | 30484 | 13295 | 1280 | 80406 | 0.0090 | 13941000 |
| 3 | 42763 | 17633 | 965 | 93464 | 0.0100 | 15483000 |
| 4 | 51581 | 21336 | 1239 | 109384 | 0.0130 | 14119000 |
| 5 | 15999 | 6380 | 263 | 62045 | 0.0060 | 14115000 |
| 6 | 26289 | 10817 | 579 | 72655 | 0.0080 | 13793000 |
| 7 | 26183 | 10624 | 481 | 74556 | 0.0080 | 13981000 |
| 8 | 39315 | 16488 | 1021 | 91162 | 0.0100 | 14799000 |
| 9 | 32550 | 14369 | 1468 | 81736 | 0.0090 | 14459000 |
| 10 | 11460 | 4627 | 166 | 58312 | 0.0050 | 14914000 |
| Average | | | | | 0.0095 | 14460600 |

From table 5 we can see that this method is very efficient by providing a 0.0095 average extraction, the features of every file remain the same from one run to another, they are also unique for each speech signal and they can be easily used as an identifier to recognize the speech.

### *3.2 WPD method implementation*

The same speech signals were treated using WPD method; table 6 shows the results of implementation

Table 6 WPD method results

| Speech # | Number of used levels | Features(packet contents at level) | | | | Extraction time(sec) | Throughput(samples per second) |
|---|---|---|---|---|---|---|---|
| 1 | 16 | -0.0082 | 0.0892 | 0.0594 | 0.0017 | 0.1690 | 1509100 |
| 2 | 15 | -0.0172 | 0.0069 | 0.0146 | 0 | 0.1500 | 836460 |
| 3 | 15 | 0.2199 | -0.0002 | 0.0788 | 0.0106 | 0.1320 | 1172900 |
| 4 | 16 | 0.0622 | 0.0976 | 0.0260 | 0 | 0.1500 | 1223600 |
| 5 | 15 | 0.0546 | -0.0168 | 0 | 0 | 0.1310 | 646500 |
| 6 | 15 | -0.0115 | -0.0082 | -0.0030 | 0 | 0.1460 | 755780 |
| 7 | 15 | 0.0775 | 0.0590 | -0.0016 | 0 | 0.1720 | 650280 |
| 8 | 15 | 0.0786 | -0.0236 | -0.0592 | -0.0120 | 0.1410 | 1049600 |
| 9 | 15 | -0.0110 | 0.0016 | 0.0371 | 0 | 0.1480 | 879240 |
| 10 | 14 | 0.0610 | 0.0484 | 0.0009 | 0 | 0.1270 | 587160 |
| Average | | | | | | 0.1466 | 931062 |

Here we can see that this method provides the same advantages as in LBPM method, but with more extraction time, to decrease the extraction time we can use the speech signal histogram. Here this histogram will be used as an initial input data set for decomposition, table 7 shows the results of implementation.

Table 7: WPDH method implementation results

| Speech # | Features(packet contents at level 6) | | | | Extraction time(sec) Including histogram calculation | Throughput(samples per second) |
|---|---|---|---|---|---|---|
| 1 | 11936 | 507 | 508 | 18929 | 0.1120 | 2277100 |
| 2 | 5155.4 | 317 | 314 | 9896.1 | 0.1060 | 1183700 |
| 3 | 7286 | 263 | 250 | 11553 | 0.1050 | 1474600 |
| 4 | 8783 | 332 | 314 | 13514 | 0.1140 | 1610000 |
| 5 | 2737.8 | 59.6 | 60.9 | 7727 | 0.0990 | 855460 |
| 6 | 44809 | 1574 | 1469 | 90068 | 0.1040 | 1061000 |
| 7 | 4467.6 | 133.3 | 126.3 | 9252.8 | 0.1010 | 1107400 |

| Speech # | Features(packet contents at level 6) | | | | Extraction time(sec) Including histogram calculation | Throughput(samples per second) |
|---|---|---|---|---|---|---|
| 8 | 6693 | 283 | 261 | 11261 | 0.1060 | 1396100 |
| 9 | 5487 | 378 | 373 | 10027 | 0.0990 | 1314400 |
| 10 | 1959.9 | 510 | 450 | 7264.1 | 0.1010 | 738310 |
| Average | | | | | 0.1047 | 1301807 |

From tables 6 and 7 we can see that WPDH method comparing with WPD gives a Speed up of 0.1466/0.1047=1.4515, which means that we can replace WPD method with WPDH method

*3.3 KMC method implementation*

The same speech files were treated using this method, but we excluded this method because of the following reasons based on the experimental results:

- The features for each speech file were not fixed and were changed from run to another.
- Some time the method failed in features extraction.
- The extraction time was very big

*3.4 FIR method implementation*

The same speech signals were treated using WPD method; table 8 shows the results of implementation.

Table 8: FIR method implementation results

| Speech # | Features | | | | Extraction time(sec) | Throughput(samples per second) |
|---|---|---|---|---|---|---|
| 1 | -0.9250 | -0.2165 | -0.3174 | 0.4703 | 0.1230 | 2073500 |
| 2 | -0.9390 | -0.1865 | -0.2856 | 0.4223 | 0.0820 | 1530100 |
| 3 | -0.8704 | -0.3159 | -0.4225 | 0.6250 | 0.1840 | 841460 |
| 4 | -0.8673 | -0.3196 | -0.4275 | 0.6322 | 0.1500 | 1223600 |
| 5 | -0.8234 | -0.3809 | -0.5047 | 0.7385 | 0.0820 | 1032800 |
| 6 | -0.8799 | -0.3067 | -0.4030 | 0.5999 | 0.0640 | 1724100 |
| 7 | -0.8564 | -0.3459 | -0.4633 | 0.6742 | 0.0720 | 1553400 |
| 8 | -0.9199 | -0.2294 | -0.3260 | 0.4855 | 0.1700 | 870530 |
| 9 | -0.9946 | -0.0062 | -0.0043 | 0.0157 | 0.0650 | 2002000 |
| 10 | -0.8217 | -0.3919 | -0.5088 | 0.7445 | 0.0600 | 1242800 |
| Average | | | | | 0.1052 | 1409429 |

Here we can see that this method provides the same advantages as in LBPM method, but with more extraction time, to decrease the extraction time we can use the speech signal histogram. Here this histogram will be used as an initial input data set for decomposition, table 9 shows the results of implementation.

Table 9: FIRH method implementation results

| Speech # | Features(packet contents at level 6) | | | | Extraction time(sec) Including histogram calculation | Throughput(samples per second) |
|---|---|---|---|---|---|---|
| 1 | -0.0247 | 0.0011 | -0.1592 | -0.0327 | 0.0700 | 3643400 |
| 2 | -0.0309 | 0.0009 | -0.1151 | -0.0108 | 0.0220 | 5703100 |
| 3 | -0.0170 | 0.0011 | -0.1495 | -0.0310 | 0.0340 | 4553800 |
| 4 | -0.0197 | 0.0012 | -0.1688 | -0.0443 | 0.0240 | 7647700 |
| 5 | -0.0028 | 0.0002 | -0.0970 | -0.0127 | 0.0160 | 5293200 |
| 6 | -0.0135 | 0.0003 | -0.1161 | -0.0161 | 0.0220 | 5015600 |
| 7 | -0.0088 | 0.0005 | -0.1265 | -0.0217 | 0.0180 | 6213800 |

| Speech # | Features(packet contents at level 6) | | | | Extraction time(sec) Including histogram calculation | Throughput(samples per second) |
|---|---|---|---|---|---|---|
| 8 | -0.0185 | 0.0013 | -0.1471 | -0.0272 | 0.0220 | 6726800 |
| 9 | -0.0357 | 0.0010 | -0.1084 | -0.0083 | 0.0220 | 5914900 |
| 10 | -0.0024 | 0.0001 | -0.0821 | -0.0089 | 0.0120 | 6214100 |
| Average | | | | | 0.0262 | 5692640 |

From tables 8 and 9 we can see that FIRH method comparing with FIR gives a Speed up of **0.1052/0.0262=4.0153**, which means that we can replace FIR method with FIRH method.

Table 10 summarizes the extraction time results for the three methods.

Table 10: Extraction time summery

| Method | Average extraction time(second) |
|---|---|
| LBPM | 0.0095 |
| WPDH | 0.1047 |
| LPCH | 0.0262 |

From table 10 we can see that LBPM has the best efficiency and it provides a speedup greater than 1 comparing with other methods as shown in table 11.

Table 11: Speedup comparisons

| | LBPM | WPDH | LPCH |
|---|---|---|---|
| LBPM | 1.0000 | 11.0211 | 2.7579 |
| WPDH | 0.0907 | 1.0000 | 0.2502 |
| LPCH | 0.3626 | 3.9962 | 1.0000 |
| Green is the best choice<br>Red is the worst choice | | | |

## 4. Conclusion

Different methods of human speech features extraction methods were studied and implemented. The obtained experimental results showed that KMC method gave the worst results and it was execluded from the comparisons. Other investigated methods are recommended for speech signal features extraction. They gave a stable, unique and fixed features for any speech file. The most recommended method is LBPM because it gaves the better speedup comparing with other methods.

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