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A Novel Approach to Spoken Arabic Number Recognition Based on Developed Ant Lion Algorithm

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ABSTRACT Intelligent spoken system is constructed to recognize numbers spoken in Arabic language by different people**.** Series of operations are performed on audio sound file as pre-processing stages. A novel approach is applied to extract features of audio files called Max Mean Log to reduce audio file dimensions in an efficient manner. Several stages of initial processing are used to prepare the file for the next step of the recognition process. The recognition process begins with the use of Antlion's advanced intelligence algorithm to determine the type of the spoken number in Arabic and later convert it to a visual text that represents the value of the spoken number. The current proposal method is relatively fast and very effective. The percentage of recognizing numbers spoken by the proposed algorithm is 99%. For 1,800 different audio files, the error rate was 1%. Additional 40 audio files were used that are different from people's original dataset. Due to an additional examination of the system and its ability to recognize the audio file, the rate of discrimination for such files was 72.5%.

KEYWORDS Pre-processing; Feature Extraction; Recognition System; Ant-lion algorithm.

I. INTRODUCTION

VOICE is an important thing for human interaction and communication. We may completely explain any basic sound by defining three characteristics: pitch, loudness (or intensity) and sound quality. Such properties correspond precisely to three physical properties: frequency, amplitude, and type of wave.

Speech recognition systems are divided into three basic directions: 1) The direction of the audible sound representation: It is based on the theory of the system of audible speech sounds, which assumes that there is a specific number of different audio units in the language, and these sound units are represented by a group of distinctive characteristics in the verbal signal or its spectrum. 2) Identification of the model: It is basically based on the use of speech models directly and without reference to any specific characteristics, as this trend depends on two main steps, which are training for speech and discrimination models for these models by matching. 3) The trend of artificial intelligence: It is a hybrid trend that combines the first direction and the second direction, as it invests the ideas and concepts of both directions. The approach to recognition is automated based on personal intelligence in visualization, analysis and decision-making to measure auditory traits [1].

Modern technologies designed to process people's voices are limited, but they are, in reality, relevant, necessary and fascinating. Since the computer is a very important in our everyday lives, this smart system has to be used in interactive and easy ways. One of these modern methods is the use of a human voice to communicate with or order it as an easy interactive method [2].

This approach is called the human computer interface because the sound is the most common form of exchanging knowledge and interaction with others. Speech recognition systems try to interpret and translate spoken words into a text that represents them [3].

These systems are used in many fields and significant life-time applications, such as: a) the smart home remote control system, b) entering the credit card numbers of a particular person via a smart application that uses the system for distinguishing voices [4] and c) the systems for identifying people by their voices [5]. Accurate systems for the automated analysis and interpretation of speech by computer are one of the major challenges facing researchers due to the variation in sound and tone from person to person and from time to time with the environmental effects and various dialects of one language and other difficulties [4, 5]. The process of speech recognition depends on many important issues, including: dataset, the methods and techniques used in the extraction of features, selecting various types of voice categories, voice representation methods, speech classification methods, evaluation of results, algorithm used, and others [4].

In previous studies, several approaches were used to translate letters or numbers spoken in different languages via converting the audio file to text. In 2010, researchers (Ahmad A. M. Abushariah et al) used the hidden Markov model (HMM) method to implement the speech recognition system, in particular to numbers spoken in English [6].

Also, in another research [7] a speech recognition system was implemented using an improved RNN training system and a lot of GPUs. This was done by researchers (Awni Hannun et al) in 2014. The researchers (Takialddin Al Smadi et al) proposed a programmatic system that uses an intelligent technology based on neural networks to identify sounds for informal users [8].

In 2018, a number of researchers (Raghav Menon, and others) organized a speech recognition system for Somalis using recurrent neural-networks RNN, long-short-term memory neural-networks (LSTMs), bi-directional long-short-term memory (BLSTMs) and time delay neural-networks (TDNNs) with convolutional neural networks (CNNs) [9].

Speech recognition system aims to convert spoken speech to written text that can be processed again in multiple forms, or offered for reading in various applications, this technology has many applications, such as desktop dictation, automated copying of broadcast newscasts, audio archives.

In this research, we suggest a new intelligent system to recognize audio file which is called SANR (Spoken Arabic Number Recognition System), it contains multi stages that will be explained in the following section.

II. FEATURE EXTRACTION AND PRE-PROCESSING

Designing a computer system to recognize some pattern (image of a certain entity, voice of a certain person, specific attribute ...), includes several stages. The pre-recognition stage involves extracting attributes from the data that was initially processed and simplified. This helps in the process of prediction, identification, optimization and data representation, because the primary data collected is typically complicated and huge. This huge data takes a lot of time when used directly in recognition systems without the use of algorithms to extract features that help to reduce dimensions and extract the best features possible thereby to reduce the time spent using them in computer systems [10].

In this research, a new algorithm method is proposed to extract the features of audio files belonging to numbers (0-9) in Arabic, the new approach was called MMLog (Max Mean Log), several stages are performed in order to extract the most important characteristics in the quickest and easiest way. The following table contains the steps of MMLog.

Table 1. Pseudo-code of Proposed MMLog

MMLog Algorithm
For each audio file do 1. Read the audio file and return the samples values for storing them in a single vector array. 2. Find the maximum value of the samples in the audio file stored in the vector [11]. 3. Normalization process is based on the following equation to get normalized sample vector: $$\text{normalized sample} = \frac{\text{original sample}}{\text{max sample}} \quad (1)$$ 4. Find the mean value for the normalized sample vector by estimating the average value in normalized sample vector according to the following equation [11]: $$\text{mean of normalized sample vector} = \frac{\sum_{i=1}^{n} \text{normalized sample}_i}{n} \quad (2)$$ 5. Simplify resulting numbers and omit the exponential values using the following equations [11]: $$x = \text{floor}(\text{Log}_{10}(m)) \quad (3)$$ $$s = \frac{m}{10^x} \quad (4)$$ 6. Find the absolute value for the results. 7. End for.

where i – represents a counter, n – is a number of elements in the normalized sample vector, and m – is the mean of normalized sample vector.

III. ANT LION ALGORITHM (ALO)

Nature-inspired algorithm simulates hunting Ant Lion and getting food. This algorithm was first suggested by Mirjalili in 2015 to solve optimization problems [12, 13].

An ant lion typically digs a conical hole in the sand, moving on a circular path that represents the top of the cone, Used a strong jaw for this reason then throws sand outside the circle, this hole is considered a hunting ants trap, hides under the cone, waiting for the tiny insects and the ants' larvae, the edge of the cone is thin and sharp enough to allow small insects going through the hole easily. Once the ant-lion know that the prey is in the hole, it begins to pull it under the soil and prey on it, it can throw the remains out of the hole and prepare the hole for the next prey note (see Fig.1) [13, 14].

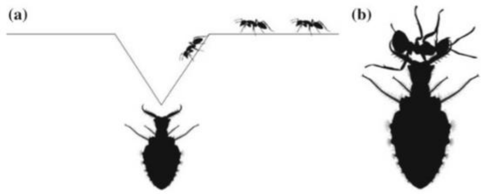


Figure 1. Ant-lion's hunting activity

This algorithm is a parallel method between exploitation and exploration, with a high probability of avoiding stagnation in the current region, thereby ensuring convergence towards an optimal solution [15]. Fig. 2 shows the flowchart of the ALO algorithm [14, 16]:

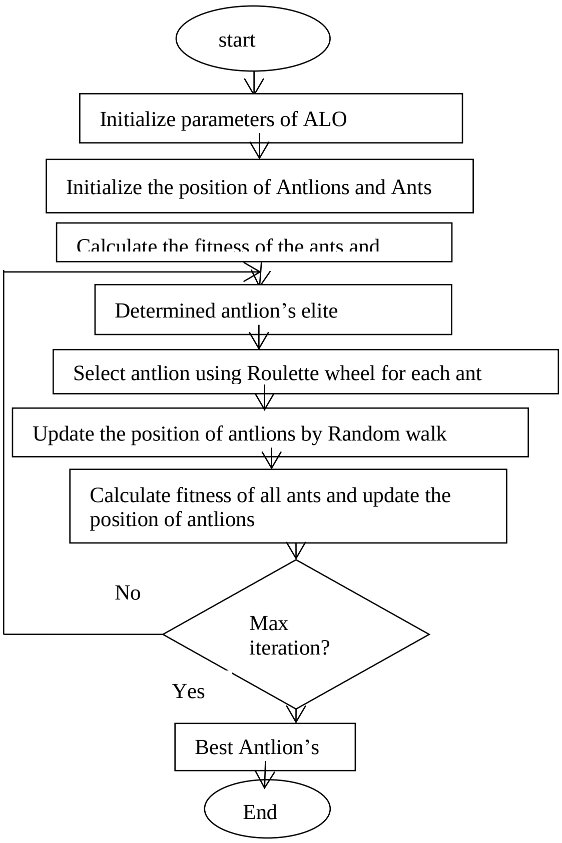


Figure 2. Flowchart of ALO

IV. DEVELOPED ANT LION ALGORITHM (DALA)

An improvement was made to the original Ant lion algorithm by adding the measurement degree of fitness to all the lions that are presented in each cycle of the population. We aim to obtain a best solution with the probability of higher convergence and ensuring a faster change from exploration to exploitation. The use of a new technique of extracting features (MMLog) has a major impact on the effectiveness of the performance of this modification and on the achievement of conclusive results.

Fitness was determined by finding the difference between the input signal after its pre-processing and extracting its characteristics from all the signals in the dataset, which also pre-processed and extracted its characteristics in the same way. That is the calculation of the difference between the ant target and all the insects of the ant lion presented in the dataset to obtain the degree of fitness for each ant lion after taking the absolute value of the result. Then apply the algorithm to get the improvement we need and find the elite in each session until the best solution is reached. Table 2 contains The Proposed Algorithm pseudo codes.

Table 2. Pseudo-code of Proposed Algorithm DALA

DALA Algorithm
1. Read the audio files and create a dataset.
2. Apply the MMLog to the dataset and Initialize the first population of ant and ant lions randomly.
3. Calculate the fitness of ants then calculate the fitness of ant lions using the equation:
$\mathrm{fitness}_i = \mathrm{ant}_i - \mathrm{antlion}_i$ (5)
4. Find the optimum ant lions and assume it as the elite.
5. While the end number of iterations is not satisfied.
6. For every ant (goal).
7. Select an ant lion using roulette wheel.
8. Update the values: c, d using equations [16, 17]:
$c^t = \frac{c^t}{I}, d^t = \frac{d^t}{I}$ (6)
9. Create the random walk and normalize it using equations [16, 17]:
$x(t) = [0, \mathrm{cumsum}(2r(t_1) - 1), \ldots \mathrm{cumsum}(2r(t_n) - 1)]$ (7)
$x_i^t = \frac{(x_i^t - p_i) * (d_i - c_i^t)}{d_i^t - p_i}$ (8)
10. Update the position of ant using the equation [14, 16]:
$\mathrm{ant}_i^t = \frac{R_A^t + R_E^t}{2}$ (9)
11. End for
12. Calculate the fitness of all ants.
13. Replace an ant lion with its corresponding ant become fitter using equation [16, 17]:
$\mathrm{antlion}_j^t = \mathrm{ant}_i^t \text{ if } \mathrm{fitness}(\mathrm{ant}_i^t) < \mathrm{fitness}(\mathrm{antlion}_j^t)$ (10)
14. Update elite if an ant lion became fitter than the elite.
15. End while.
16. Return elite.

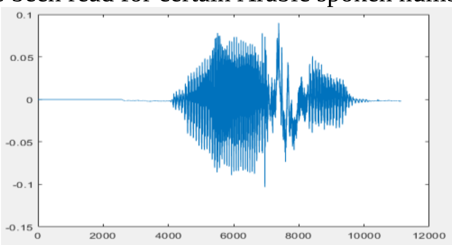
where p_i represents the minimum random walk of the variable *i*, c_i^t represents maximum random walk in variable *i*, it is the minimum of *i* variable at iteration *i*, and it indicates the maximum of *i* variable at iteration *t*. While R_A^t is a random walk around antlion which is selected by roulette wheel at the iteration *t*, R_E^t is a random walk around elite at iteration *t*, and ant_i^t represents position of *i* ant at iteration *t*. $\mathrm{Antlion}_j^t$ shows the position of selected *j* antlion at the iteration *t*, *t* is the current iteration and ant_i^t represents the position of the *i* ant at iteration *t*.

V. EXECUTION OF SPOKEN ARABIC NUMBER RECOGNITION SYSTEM (SANR)

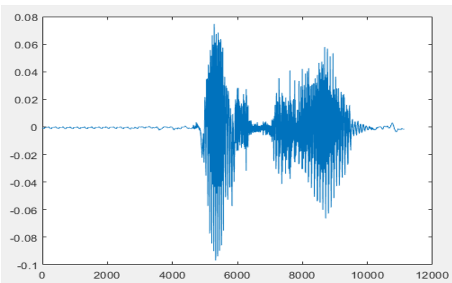
To create the dataset, ten people were involved. Every person counts Arabic numbers from zero to nine, a different number each time to get 176 audio files of each number. The system was implemented as follows:

A. READ ALL AUDIO FILES AND RETURNED SAMPLES

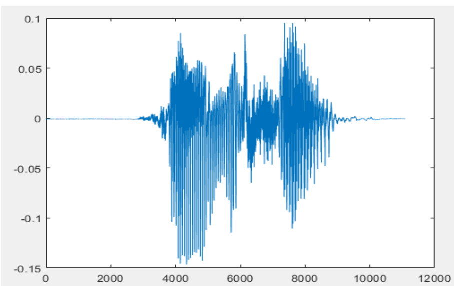
The following figure shows examples of some audio files that have been read for certain Arabic spoken numbers.



a) Number1 (the voice of word“wahed”)



b) Number9(the voice of word “tessaa”)



c) Number5 (the voice of word “khamsa”)

Figure 3. a), b), c) Some of audio files

B. IMPLEMENTATION OF THE MMLOG ALGORITHM TO EXTRACT THE FEATURES OF ALL SAMPLE FILES

Table 3 shows some results of the algorithm for the audio files of the Arabic numbers.

Table 3. Some results of MMLOG algorithm for audio files of the Arabic numbers

No. 0	No. 1	No.3
1.591531996998950	1.675111173313256	7.105978849658982
1.178204383266864	1.422542626198523	5.646109484815721
1.046572087569880	2.204805563656399	6.734852434095741
1.043956374686484	1.854866263680750	5.883517692152588
1.394772413094738	1.854698841867980	6.607177630168002
1.380908756049706	1.540584808241369	5.566751258718044
1.194752659562508	1.673568151194718	6.784422285376550
1.267001819624418	1.943898650045233	8.814964445014487
1.109296794576016	1.414086580593309	6.715122288388536
1.072248504313420	1.399484862344828	4.947461333903068
No. 3	**No. 4**	**No. 5**
2.083557165209221	1.516166507337233	2.083557165209221
2.059309427388306	1.501301241494467	2.059309427388306
2.483989647821078	1.603020386713853	2.483989647821078
2.470152762714920	1.764171508322895	2.470152762714920
2.097769060665392	1.503680299622348	2.097769060665392
2.075301542502548	1.619106118283359	2.075301542502548
2.899711598746140	1.629195532623529	2.899711598746140
2.851314771740139	1.634333138273531	2.851314771740139
2.045916246239761	1.177596224735220	2.045916246239761
2.328357743872828	1.330038117327843	2.328357743872828
No. 6	**No. 7**	**No. 8**
3.651711279678248	8.755239451630901	2.634340042879335
3.667406728864992	8.938218658684649	2.615707526602133
3.651711279678248	8.938218658684649	1.820749753186729
4.166070777621512	8.838148248804643	2.644221644652070
3.177027188884899	8.274473071274060	2.643829397737072
3.210670530834820	5.308472161945075	2.594181108060647
4.027678157815013	6.641653110152736	2.606105384734046
4.019927165627325	6.034819702721998	2.168403347554957
4.657432010025388	8.302001364547086	2.197705374562739
4.688199170454793	9.228968162139527	3.206471660187393
	No. 9	
	1.784950588143625	
	1.979858092859883	
	1.789702040898691	
	1.720164168980404	
	1.979481552797582	
	1.752889709346190	
	2.053349686847156	
	1.705998981666284	
	1.176265601076257	
	1.480505696500991	

VI. RESULTS AND DISCUSSION

In this section, the results are presented along with the discussion about performing multiple experiments with a group of people. All the numbers used in the research are pronounced separately for many times, and each number is spoken 176 times by a different number of people. In addition, the results are evaluated by using a set of measures to determine the degree of accuracy of the system for the identification and recognition of spoken number.

A. CRR MEASURE

This measure represents the system's ability to correctly identify and recognize the type of spoken number of the numbers in the dataset. CRR measure is represented in the following equation [18]:

$$\text{Correct Recognition Ratio(CRR)} = \frac{\textit{Number Of Audio Files Correctly Recognized}}{\textit{Number Of All Audio Files}} \tag{11}$$

B. WRR MEASURE

Numerical value of WRR measure reflects negative system discrimination. In other words, it is difficult to distinguish those audio files that may contain high distortion or that are far from the target as a numerical value. The lower value of this measure therefore means a higher performance of the system. WRR measure is shown in the following equation [19]:

$$\text{Wrong Recognition Ratio(WRR)} = \frac{\text{Number Of Audio Files Wrongly Recognized}}{\text{Number Of All Audio Files}} \tag{12}$$

C. RECALL MEASURE

It gives a calculation of the total results that are highly relevant to the target and have been correctly recognized. It represents the percentage of positively recognized audio files that have been successfully recognized by the ratio of audio files that have been identified as correct and positive in addition to the percentage of audio files which have been incorrectly and negatively identified which represent audio files which should have been correctly distinguished within the system. Recall measure is explained in the following equation [20]:

$$\text{Recall} = \frac{\text{True positive(TP)}}{\text{True Positive(TP)+False Negative(FN)}} \tag{13}$$

D. ACCURACY MEASURE

It is a method to measure the accuracy of the system recognition for audio files that are unfamiliar to the system. This means the ratio of the number of positively files recognized to the ratio of the recognized correct and positive files with the audio files that have not been recognized within the system. It is a good thing for the system in the event that they are not distinguished, they are strange files and do not correspond to the numbers needed to be recognized, so rejecting them was a correct decision. The output of this scale is numerical value and it is better to identify sounds that are not related to the numbers used in the system. Recall measure is explained in the following equation [21]:

$$\text{Accuracy} = \frac{\text{True positive(TP)}}{\text{True Positive(TP)+False Positive(FP)}} \tag{14}$$

E. FNR MEASURE

It indicates that the foreign audio files were correctly and negatively distinguished and that they were not supposed to be recognized by the system and were expected to reject it. As this measure deals with the percentage of audio files that are rejected negatively and which represent strange files, but are dependent on the same numbers used in the system in addition to the percentage of files that were distinguished correctly and positively, the output value of this measure is a number, and the lower value means better results. The following equation represents FNR measure [21]. Table 3 shows the results of SANR proposed system.

$$\text{False Negative Ratio(FNR)} = \frac{\text{False Negative(FN)}}{\text{True Positive(TP)+False Negative(FN)}} \tag{15}$$

After testing the proposed SANR program on a collection of specific audio files for a group of people of different age groups (15 years to 50 years), the results proved that the proposed algorithm achieved a very high CRR of 99% for “Familiar” sound files in a database (The number of files reached 1,800 audio files) that was used to initialize and build the proposed algorithm. Also, the value of Recall is 99%, and Accuracy is 100%, while FNR is 1%, these results indicate the efficiency of the proposed algorithm especially after coming up with a suggestion for the MMLOG algorithm to extract the features, where after using it, good qualities were obtained that help greatly in the process of distinguishing. At the same time, this method significantly reduced the size of the audio file, it greatly reduced the amount of data that the developed AntLion algorithm would handle, as shown in Table 3.

It also added modifications that were developed from the results of the traditional terms AntLion algorithm where good results were obtained as shown in Table 4.

And after conducting additional experiments on the unfamiliar database (40 audio files), CRR reached 72.5%, and Accuracy 93%, these values are good and satisfactory results for any recognition, especially when dealing with sound files.

Table 4. Results of SANR proposed system

No.	1	2
No. of audio files	1800	40
Types of audio files	Familiar	Unfamiliar
TP	1782	27
FP	0	2
FN	18	11
CRR	99%	72.5%
WRR	1%	27.5%
Recall	99%	71%
Accuracy	100%	93%
FNR	1%	28%

VII. CONCLUSIONS

The results show that the algorithm operates well with audio files and is able to identify the numerical values spoken in Arabic in an excellent way. Using the proposed method of extracting features MMLog, involves the implementation of mathematical and logarithmic processes with simple exponential effects. High distinguishing results are obtained. In addition to the use of the proposed modification of the Ant Lion algorithm, the probability of random convergence and improvement increased. Based on our results, we consider that the ability of the algorithm to achieve the goal is quick and accurate, because the time of implementation does not exceed 9 seconds and the accuracy of the recognition was 99 per cent. This indicates that the algorithm is very close to exploitation, particularly in the initial stages of iteration, and the random walking from time to time helps the algorithm in the exploration process. Thus, maintaining the necessary balance between exploration and exploitation is essential in order to quickly find a solution. Finally, the proposed and developed algorithms in this research can be modified to recognize audio files that contain letters or words so that different audio clips contribute to the process of interaction between the computer and the human being.

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