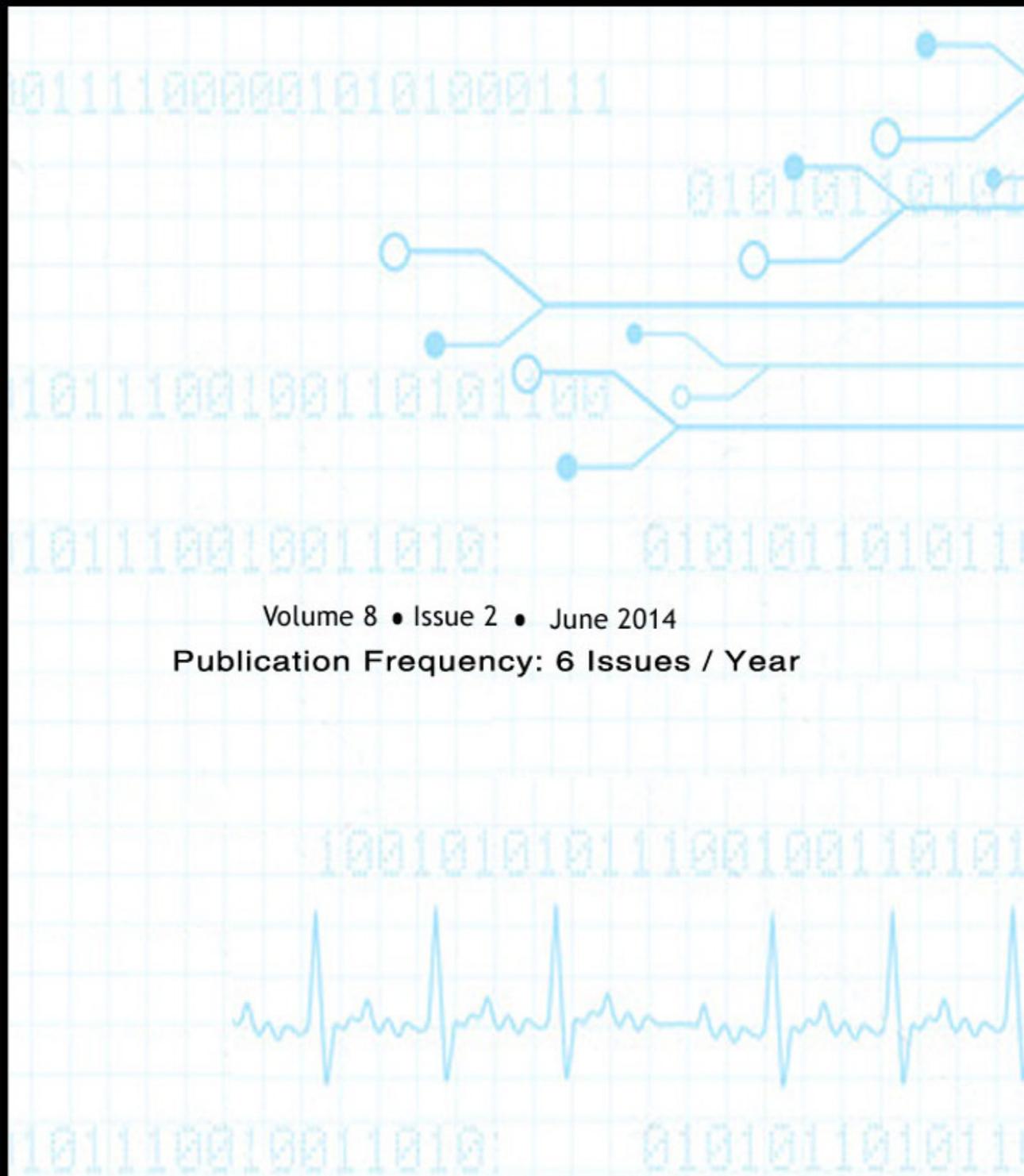


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SIGNAL PROCESSING (SPIJ)

AN INTERNATIONAL JOURNAL

ISSN : 1985-2339



Volume 8 • Issue 2 • June 2014

Publication Frequency: 6 Issues / Year

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SIGNAL PROCESSING: AN INTERNATIONAL JOURNAL (SPIJ)

VOLUME 8, ISSUE 2, 2014

**EDITED BY
DR. NABEEL TAHIR**

ISSN (Online): 1985-2339

International Journal of Computer Science and Security is published both in traditional paper form and in Internet. This journal is published at the website <http://www.cscjournals.org>, maintained by Computer Science Journals (CSC Journals), Malaysia.

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SIGNAL PROCESSING: AN INTERNATIONAL JOURNAL (SPIJ)

Book: Volume 8, Issue 2, June 2014

Publishing Date: 15-06-2014

ISSN (Online): 1985-2339

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Published in Malaysia

Typesetting: Camera-ready by author, data conversion by CSC Publishing Services – CSC Journals, Malaysia

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Mohamed Khalil Krichi, Adnan Cherif

The Arabic Speech Database: PADAS

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Abstract

This work describes a construction of PADAS “Phonetics Arabic Database Automatically segmented” based on a data-driven Markov process. The use of a segmentation database is necessary in speech synthesis and recognizing speech. Manual segmentation is precise but inconsistent, since it is often produced by more than one label and require time and money. The MAUS segmentation and labeling exist for German speech and other languages but not in Arabic. It is necessary to modify MAUS for establish a segmental database for Arab. The speech corpus contains a total of 600 sentences recorded by 3 (2 male and 1 female) Arabic native speakers from Tunisia, 200 sentences for each.

Keywords: HTK, MAUS, Phonetic Database, Automatic Segmentation.

1. INTRODUCTION

Many researches such as automatic speech recognition or speech synthesis are now based on database e.g. English [1, 2, 3 and 4]. For obtaining a good result, the database must be balanced, segmented and reduce the noise (noise in step of record)In order to produce a robust speaker-independent continuous Arabic, a set of speech recordings that are rich and balanced is required. The rich characteristic is in the sense that it must contain all the phonemes of Arabic language. It must be balanced in preserving the phonetics distribution of Arabic language too.

This set of speech recordings must be based on a proper written set of sentences and phrases created by experts. Therefore, it is crucial to create a high quality written (text) set of the sentences and phrases before recording them. Any work based on the learning step requires a database to learn the system and then evaluate it. They are a several international databases in field of speech such as TIMIT which was developed by DARPA Committee for American English. And we also find other databases in different known languages, such as French and German, and unknown, as Vietnamese and Turkish.

For Arabic, we have not found a standard database, but we still found a few references. KACST [5] database developed by the Institute of King Abdul -Aziz in Saudi Arabia.

1.1 KACST

Indeed KACST created a database for Arabic language sounds in 1997. This database was to created the least number of phonetically rich Arabic words. As a result, a list of 663 phonetically rich words containing all Arabic phonemes.

The purpose is used for Arabic ASR and text-to-speech synthesis applications.

KACST produced a technical report of the project "Database of Arabic Sounds: Sentences" in 2003. The sentences of Arabic Database have been written using the said 663 phonetically rich words. The database consists of 367 sentences; 2 to 9 words per sentence.

The purpose is to produce Arabic phrases and sentences that are phonetically rich and balanced based on the previously created list of 663 phonetically rich words [6].

1.2 ALGASD

ALGERIAN ARABIC SPEECH DATABASE (ALGASD) [7] developed for the treatment of Algeria Arabic speech taking into account the different accents from different regions of the country. Unavailability and lack of resources for a database audio prompted us to build our own database to make the recognition of numbers and operations of a standard calculator in Arabic for a single user. We made 27 recordings of 28 vocabulary words.

Database is the most important tool for multiple domains as speech synthesis or speech recognition. to provide database a interesting and contains all the acoustic units must have all the possible linguistic combinations .The quality of the final result of the synthesis is directly dependent on the quality of recordings made during the development of the acoustic units therefore a filtering step dictionary is mandatory.

The implementation stages can be summarized as follows:

- a) The choice of dictionary (set of sentences contains several examples of phonemes.).
- b) Sound recording expressions.
- c) Noise reduction.
- d) Segmentation.

2. ARABIC LANGUAGE

Statistics show that it is the first language (mother-tongue) of 206 million native speakers ranked as fourth after Mandarin, Spanish and English [8].The Arabic language is a derivational and inflectional language. The original Arabic is the language spoken by the Arabs. In addition, it is the sacred language of the Koran and Islam. Because the spread of Islam and the spread of the Qur'an, the language became a liturgical language. It is spoken in 22 countries, while the number of speakers is more than 280 million [9].

2.1 Alphabet

2.1.1 Consonants

The Arabic alphabet consists of twenty eight consonants (see Table 1) basic, but there are authors who treat the letter (alif) as the twenty –ninth consonant. The (alif) behaves as a long vowel never found as consonant of the root.

Vowels are not as consonants, they are rarely noted. They are written only to clarify ambiguities in the editions of the Koran or in the academic literature. Indeed, vowels play an important role in the Arabic words, not only because they remove the ambiguity, but also because they give the grammatical function of a word regardless of its position in the sentence. In other words, vowels have a dual function: one morphological or semantic and the other is syntactic. Arabic has two sets of vowels, the short one and the other long.

2.1.2 Short Vowels

The short vowels ($\overset{\text{◌}}{ا}$, $\overset{\text{◌}}{ي}$, $\overset{\text{◌}}{و}$) are added above or below consonants. When the consonant has no vowel, it will mark an absence of vowel represented in Arabic by a silent vowel ($\overset{\text{◌}}{◊}$).

2.1.3 Long Vowels

Long vowels are long letters, they are formed by a brief vowels and one of the following letters (ي, و, ا)

2.2 The Diacritics

Short vowels are represented by symbols called diacritics (see Figure 5). Three in number, these symbols are transcribed as follows:

- The Fetha [a] is symbolized by a small line on the consonant (َ / ma /)
- Damma the [u] is symbolized by a hook above the consonant (ُ / mu /)
- The kasra [i] is symbolized by a small line below the consonant (ِ / mi /)
- A small round o symbolizing Sukun is displayed on a consonant when it is not linked to any vowel.

2.2.1 The Tanwin

The sign of tanwin is added to the end of words undetermined. It is related to exclusion with Article determination placed at the beginning of a word. Symbols tanwin are three in number and are formed by splitting diacritics above, which results in the addition of the phoneme / n / phonetically:

[an]: (َ / Alan /)

[un]: (ُ / Alun /)

[in]: (ِ / Alin /)

2.2.2 The Chadda

The sign of the chadda can be placed over all the consonants non initial position. The consonant which is then analyzed receives a sequence of two consonants identical:

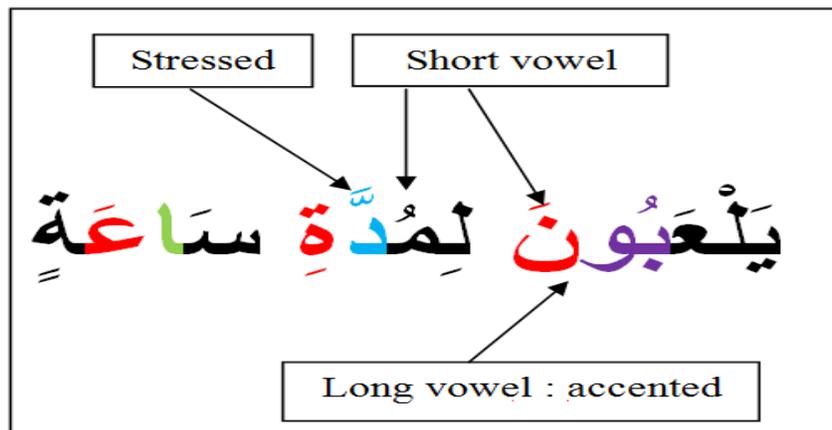


FIGURE 1: Example of a sentence / jalAlabuuna limuddati saAltin / ("They play for an hour").

Graphemes	symbol	Graphemes	symbol	Graphemes	symbol	Graphemes	symbol
ء	?	ر	r	غ	g	ي	j
ب	b	ز	z	ف	f	اَ	a
ت	t	س	s	ق	q	اِ	a:
ث	T	ش	S	ك	k	اِي	i
ج	Z	ص	s'	ل	l	اِي	i:
ح	X	ض	d'	م	m	اُ	u
خ	x	ط	t'	ن	n	اُو	u:
د	d	ظ	D'	ه	h		
ذ	D	ع	?'	و	w		

TABLE 1: Arabic consonant and vowels and their SAMPA code.

3. BALANCED SELECTION OF ARABIC WORDS

The syllabic structures in Arabic are limited in number and easily detectable. Every syllable in Arabic begins with a consonant followed by a vowel which is called the nucleus of the syllable. Short vowels are denoted by (V) and long vowels are denoted by (VV). It is obvious that the vowel is placed in the second place of the syllable. These features make the process of syllabification easier. Arabic syllables can be classified either according to the length of the syllable or according to the end of the syllable. Short syllable occur only in CV form, because it is ending with a vowel so it is open. Medium syllable can be in the form of open CVV, or closed CVC. Long syllable has two closed forms CVVC, and CVCC. Arabic words are composed at least by one syllable; most contain two or more syllables. The longest word is combined of five syllables. Table II illustrates Arabic syllables. Some of the Arabic words are spelled together forming new long words with 6 syllables like (يَأْكُلُونَهُ), or 7 syllables like (يَسْتَقْبِلُونَهُ). There exist a few Arabic data suitable for HMM-based synthesis, which should ideally include a large number of Arabic databases from a single speaker and corresponding phonetic transcriptions.

Syllable	Arabic example		English meaning
cv	لِ	li	to
cvv	فِي	fii	in
cvc	قُلْ	qul	say
cvcc	بَحْرُ	bahr	sea
cvvc	مَالٌ	maAl	money
cvvcc	زَارَ	zaArr	visit

TABLE 2: Arabic Syllables Types.

3.1 Corpus Description

Creating phonetically rich and balanced text corpus requires selecting a set of phonetically rich words, which are combined together to produce sentences and phrases. These sentences and phrases are verified and checked for balanced phonetic distribution. Some of these sentences and phrases might be deleted and/or replaced by others in order to achieve an adequate phonetic distribution [10]. The corpus, which we used to build our database, is composed of 200 sentences, with an average of 5 words per sentence. These sentences contain 1000 words, 2600 syllables, 7445 phonemes including 2302 short vowels and long vowels. These sentences were read at an

average speed (from 10 to 12 phonemes/second) by Tunisian speakers, two male and a female. They were sampled at 16 KHz with 16 bits per sample.

3.2 Corpus Analysis

We have carried out a statistical study of our corpus. Table 3 shows the results of this study. We can note the following results:

- ❖ The short vowel [a] and the long vowel [a:] appear with a frequency of 17%, followed by vowels [i] and [i:] with an occurrence frequency of 14.3%. The vowels [u] and [u:] represent 7%.
- ❖ The occurrence of the vowel (short and long) is about 37%.
- ❖ The most frequent Arabic consonants are: [ʔ] (15%), [n] (6.66%), [l] (6.63%), [m] (5.59%), etc.

Consonant and vowels	Phoneme Repetitions	%
ʔ	523	13,34%
b	102	2,60%
t	92	2,35%
T	70	1,79%
x	19	0,48%
/X	20	0,51%
G	35	0,89%
d	39	0,99%
D	40	1,02%
r	102	2,60%
z	35	0,89%
s	48	1,22%
S	73	1,86%
s'	18	0,46%
d'	24	0,61%
t'	19	0,48%
D'	24	0,61%
ʔ'	61	1,56%
g	23	0,59%
f	61	1,56%
q	61	1,56%
k	80	2,04%
l	260	6,63%
m	219	5,59%
n	261	6,66%
h	123	3,14%
w	51	1,30%
j	80	2,04%
a	400	10,20%
a:	254	6,48%
i	400	10,20%
i:	28	0,71%
u	252	6,43%
u:	23	0,59%
total	3920	100%

TABLE 3: Occurrence Frequency (%) of Arabic Consonants and Vowels.

3.3 Noise Reduction

Degradation of signals by noise is an omnipresent problem [11]. In almost all fields of signal processing the removal of noise is a key problem. The wavelet transform is striking for its great variety of different types and modifications. A whole host of different scaling and wavelet functions (or scaling and wavelet coefficients) provide plenty of possible adjustments and regulating variables [12]. The audio recordings were noisy with a continuous background noise. Our goal is to reduce this undesirable component. Figures 2 and 3 shows the time signal before and after filtering for a particular audio file. We note in particular that the zone of silence highlighted is closer to zero in the filtered signal in the original signal release.

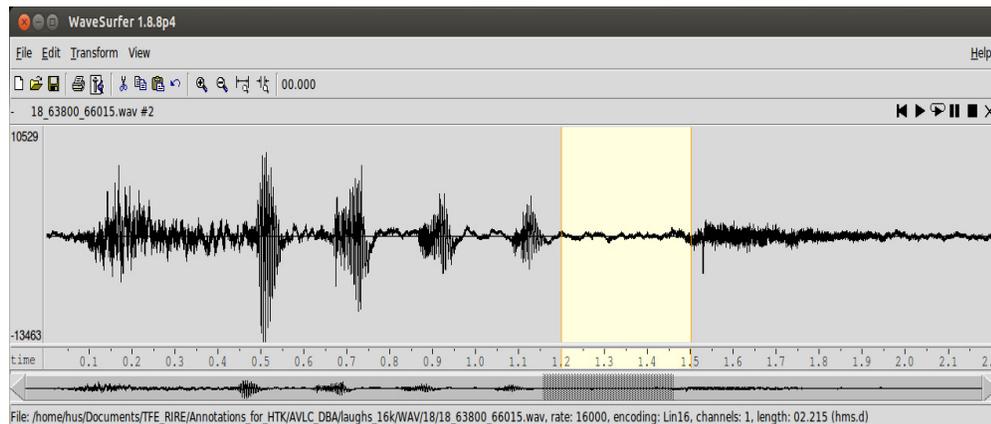


FIGURE 2: Example for Original Speech of Database File.

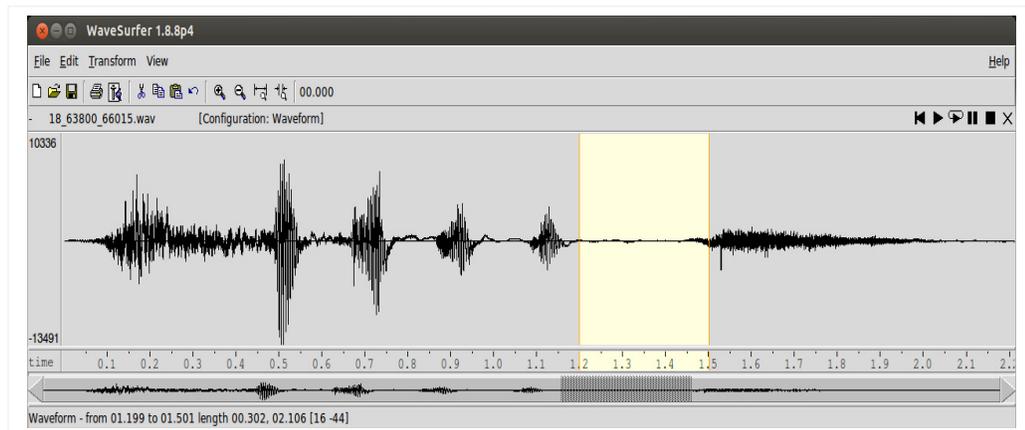


FIGURE 3: Example for the Same Speech Denoising.

3.4 Automatic Segmentation

Nowadays, the Practical applications of automatic S&L are implemented as a statistical search for a S&L \hat{k} in a space Ψ of all possible S&Ls, which can be formulated as [13,14]:

$$\hat{k} = \arg \max_{k \in \Psi} P(k | O) = \arg \max_{k \in \Psi} \frac{P(k)p(O | k)}{P(O)} \quad (1)$$

Where, O is the acoustic observation on the corresponding speech signal. The MAUS system

models $P(k)$ for each recording O . Each path from the start node to the end node represents a possible $k \in \psi$ and accumulates to the probability $P(k)p(O|k)$ which is determined by HMMs for each phonemic segment and a simple Viterbi search through the graph yields the maximal $P(k)p(O|k)$.

The 'Munich Automatic Segmentation' (MAUS) system developed by Department of Phonetics, University of Munich, For more details about the MAUS method refer to [15], [16] and [17].

The purpose is analyzing a spoken utterance. Indeed, input a speech wave and some orthographic form of the spoken text. The text is parsed into a chain of single words (punctuation marks are stripped) and passed to a text-to-phoneme algorithm, which is either rule-based or a combination of lexicon lookup and fallback to the rule-based system.

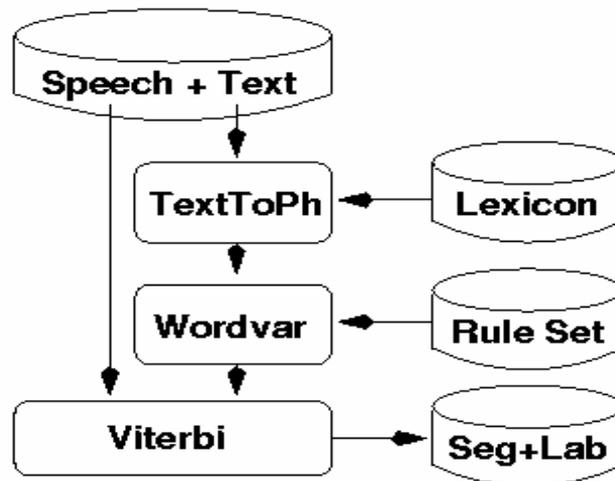


FIGURE 4: Processing in MAUS.

3.5 Corpus Segmentation and Labeling

Our continuous speech corpus was segmented and labeled with automatic procedure "MAUS". This software requires as input a file in format "wav" of the sentence to be segmented, and a text file containing the phonetic transcription of the same sentence. A phonetic file (.par). This file consists of the list of sentence phonemes with their prosodic characteristics.

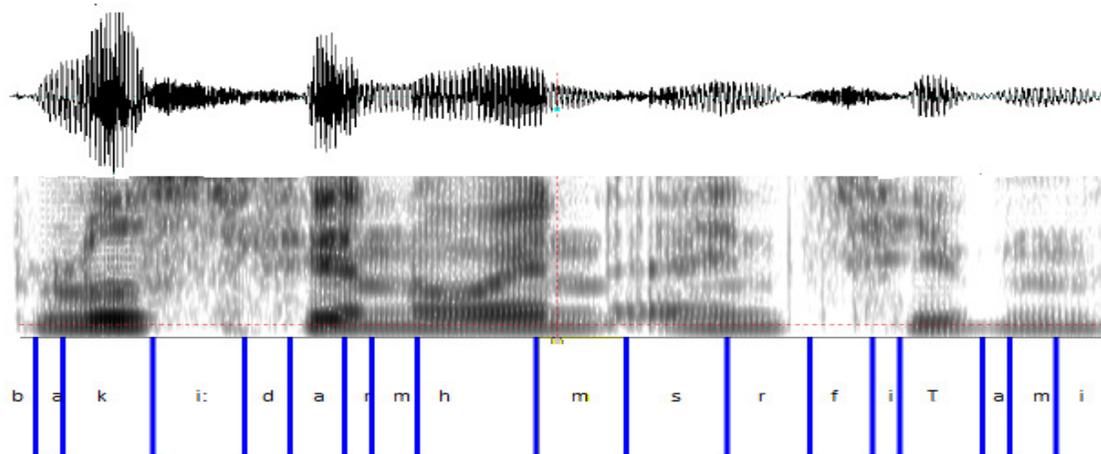


FIGURE 5: Example MAUS segmentation and labeling taken from the Arabic corpus with SAMPA code.

4. RESULT AND EVALUATION

4.1 Result

A database is a collection of accumulated documents. our database defines as follow:

The files (.wav), four files of transcription (txt, word, phn, textGrid) exist for each sentence of the corpus, which contains respectively:

- The text of the marked sentence (.txt) ;
- The associated time aligned word transcription (.word) ;
- The associated time aligned phonetic transcription (.phn) ;
- Temporal description of each phoneme; start time and end time (.textGrid).

```
File type = "ooTextFile"
Object class = "TextGrid"

xmin = 0
xmax = 2.010000
tiers? <exists>
size = 1
item []:
  item [1]:
    class = "IntervalTier"
    name = "MAU"
    xmin = 0
    xmax = 2.010000
    intervals: size = 18
    intervals [1]:
      xmin = 0.000000
      xmax = 0.320000
```

FIGURE 6: Temporal description of each phoneme; start time and end time.

4.2 Evaluation

In total, 600 sentences were segmented, 400 sentences for the two speakers (male, 200 sentences for every one), 200 sentences for the third speaker (female). For each segmented 200 sentences, we randomly selected 10 sentences for segmented manually. To do this, we need 6 students in our research laboratory, two for each 10 sentences. The results are summarized in the following table:

speaker	Manual segmentation	Automatic segmentation
First male speaker	99%	94%
second male speaker	99%	94.4%
female speaker	99%	94.1%

TABLE 4: Evaluation Result.

5. CONCLUSION

This paper reports our work towards developing the PADAS «Phonetic Arabic Database Automatically Segmented» based on rich phonetic and balanced speech corpus, which is automatic segmented with the MAUS system. This work includes creating the rich phonetic and balanced speech corpus; building an Arabic phonetic dictionary, reducing noise by wavelet method and an evaluation of the automatic segmentation. The current release of our database contains 1 female and 2 male voices. The purpose of this work is to build a database to be used in all area of Speech processing. This variety is useful when used in speech synthesis or speech recognition.

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