An Attempt to Create Speech Synthesis Model That Retains Lombard Effect Characteristics

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Abstract: The speech with the Lombard effect has been extensively studied in the context of speech recognition or speech enhancement. However, few studies have investigated the Lombard effect in the context of speech synthesis. The aim of this paper is to create a mathematical model that allows for retaining the Lombard effect. These models could be used as a basis of a formant speech synthesizer. The proposed models are based on dividing the speech signal into harmonics and modeling them as the output of a SISO system whose transfer function poles are multiple, and inputs vary in time. An analysis of the Lombard effect of the synthesized signal is performed on the noise residual. The synthesized signal residual is described by vectors of acoustic parameters related to the Lombard effect. For testing the performance of the created models in various noise conditions two classifiers are employed, namely kNN and Naive Bayes. For comparison of results, we created models of sinusoids based on frequency tracks. The results show that a model based on the residual sinewave sum demonstrates the possibility of retaining the Lombard effect. Finally, future work directions are outlined in conclusions.

1 INTRODUCTION

Even though researchers and engineers try to automate speech recognition and synthesis at least for half of a century, the progress in this field is below expectations. This especially concerns speech synthesis and speech in noise production and perception. Both research areas require a thorough analysis of individual spoken elements, carried out individually for languages. Analysis of speech in a noisy environment is an important aspect to deepen knowledge which is still missing. In the presence of noise, production of speech is modified. One of the most prominent effects of noise on speech production is called the Lombard effect, named after the French oto-rhino-laryngologist (Zollinger and Brumm, 2011), who discovered "the symptom of the raised voice", i.e., vocal effort expended due to noise. Lombard determined that in order to improve the audibility, the speakers increased the level of their voice when they were in intense, adverse noise conditions, for example in noisy environments, in restaurants, etc. He also stated that the speakers were not aware of this effect. The effect was decided to be used to diagnose the degree of deafness, as well as to reveal people who simulate hearing problems.

By definition, the Lombard effect (LE) is defined as the unintended tendency of the interlocutor to increase the level of speech in noise conditions in order to improve audibility and intelligibility (Kim, Davis, 2014). It was shown that LE manifests itself in many other speech variables than intensity only. LE causes changes in frequencies of the fundamental and formants, duration of vowels, signal spectrum slope flattening (Brum and Zollinger, 2018), etc. (Folk and Schiel, 2011; Godoy et al., 2014; Garnier et al., 2006). The Lombard effect was and is extensively researched (Boril and Pollák, 2005; Boril and Hansen,

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2010; Kim J. and Davis, 2014; Garnier and Henrich, 2013; Kleczkowski et al., 2017; Krishnamurthy and Hansen, 2009; Van Summers et al., 1988; Vlaj and Kačič, 2011), contrarily synthesis of LE is visible in the literature to less extent.

The speech with the Lombard effect has been studied extensively in the context of speech recognition (Vlaj and Kačič, 2011; Marxer et al., 2018). However, less attention has been paid to speech with Lombard effect synthesis. One of the reasons for this is that recording a large database of units that would be used in unit selection or concatenative synthesis is an extremely complex task. The authors of papers (Huang et al., 2010; López et al., 2017) propose a speech transformation approach to mimic this Lombard effect for improving the intelligibility of speech in noisy environments. These research has tended to focus on speaking style conversion from normal to Lombard speech, rather than creating speech models with Lombard effect. To our knowledge, only a few studies (Raitio et al., 2011; Suni et al., 2013) describe the speech synthesis with the Lombard effect. These studies are, however, based on hidden Markov models (HMMs). It is well known that the naturalness of synthetic speech given through HMM-based synthesis system is not as good as that of the unit-selection or formant speech synthesizers.

The goal of this research is to create the models of speech which retain the Lombard effect. Such a model may help to synthesize these speech variables that may facilitate communication and perception in noisy adverse environments. These models could be used as a basis of formant synthesizer, which has advantages against other synthesizers. It produces sufficiently intelligible speech even at high speed, and most importantly, it can control prosody aspects of the synthesized speech.

The main focus of most of the scientific papers which cover speech synthesis is on quality of the synthesized language. In this work, we look at the synthesis of speech from another perspective. Therefore, we propose a speech model which reconstructs the Lombard effect in the synthesized speech.

The objective of this research is focused on vowel and semivowel speech phonemes analysis with the Lombard effect. Since the character of vowel and semivowel sounds is periodic, according to Fourier theory, these sounds can be expanded into the sum of harmonics. It is a well-known fact that information, especially that, which is not located in the harmonic peaks is not reproduced in the synthesized signal. Therefore, for harmonic modeling, a harmonic generator based on SISO (Single-Input and Single-Output) system proposed by Korvel *et al*. (Korvel et al., 2016) is used in this paper. In order to distinguish whether the synthesized signal covers information concerning Lombard effect, an analysis of noise residual of regular speech and that with Lombard effect is performed. For a comparison of results, we created models of sinusoids based on frequency tracks (Serra, 1997).

The paper is organized as follows: first, the basis of the harmonic model is recalled. In the next Section, the main assumptions regarding sinewave modeling based on frequency trackers are given. Then, noise residual analysis technique is described based on parametric description and examination of the extracted speech parameters. The experimental part consists in speech recordings, extraction of the vowel and consonant phonemes data from the recordings of utterances in the noise conditions, and the analysis of natural and synthesized speech uttered in the absence/presence of noise. Individual parameters for which the analyses were conducted are specified. Additionally, speech classification results are presented employing two well-known classifiers, i.e., kNN (k-Nearest Neighbors) and Naive Bayes. Finally, conclusions are derived on the basis of the results obtained. Also, future development for synthesizing Lombard effect in speech is outlined.

2 HARMONIC MODELING BASED ON SISO SYSTEM

Speech signals of vowels and semivowels are periodic. Mathematically, a periodic signal can be approximated by the sum of harmonics:

$$x(t) = \sum_{k=1}^{\infty} a_k \sin(2\pi k f_0 t + \varphi_k) \tag{1}$$

where $x(t)$ is the phoneme signal, f_0 is the fundamental frequency of the signal, a_k refers to the amplitude of k^{th} harmonic and φ_k refers to the phase of the k^{th} harmonic.

Due to the fact that very high frequencies do not affect the sound of the speech signal, the infinity symbol in Eq. (1) can be changed to a finite number of sinewaves (denoted by K). Also, it should be noted, that the periods are not completely identical. Therefore, in order to get a natural sounding, it is assumed that the harmonic amplitudes and the fundamental frequency are functions of time. Therefore Eq. (1) should be rewritten to contain respectively: $a_k(t)$ and $\varphi_k(t)$. For a harmonic generation, we used the model proposed by Korvel *et al.* (Korvel et al., 2016). The harmonic is given as the output of a SISO system whose transfer function poles are multiple and inputs vary in time. The impulse response of such a system is the 4th order quasipolynomial:

$$h_k(n) = e^{-\lambda_k n\Delta t}\sum_{m=1}^{4} a_{km}(n\Delta \mathrm{t})^{m-1}\sin(2\pi k f_k\, n\Delta \mathrm{t} + \varphi_{km}) \qquad (2)$$

where n is the discrete time, Δt – refers to the sampling period, $k = 1, \dots, K$ (K – the number of harmonics), λ_k is the damping factor, a_{k1}, a_{k2}, a_{k3}, a_{k4} are amplitudes and φ_{k1}, φ_{k2}, φ_{k3}, φ_{k4} – denote phases.

The edges of k^{th} harmonic filter are calculated by the formula:

$$P_{star} = ((k-1)+0.5)f_0 \qquad (3)$$

$$P_{end} = (k+0.5)f_0 \qquad (4)$$

where f_0 is the fundamental frequency of the analyzed phoneme.

The parameters of quasipolynomial are obtained using Levenberg method and component-wise optimization with deconvolution (Slivinskas and Simonyte, 2009).

A harmonic changing over time is obtained by using the system inputs with time-varying amplitudes and slightly varying periods.

Let T be a vector consisting of the phoneme periods:

$$T = [T_1, T_2, \dots, T_M] \qquad (5)$$

where M is the number of periods of the phoneme. The number of periods of the k^{th} phoneme is close to the number $k \cdot M$. Therefore the vector consisting of the k^{th} harmonic periods can be expressed by the following formula:

$$T_k = \left[\underbrace{\frac{T_1}{k},\dots,\frac{T_1}{k}}_{k},\underbrace{\frac{T_2}{k},\dots,\frac{T_2}{k}}_{k},\dots,\underbrace{\frac{T_M}{k},\dots,\frac{T_M}{k}}_{k}\right] = [T_{k,1},\ T_{k,2,},\dots,T_{k,kM}] \qquad (6)$$

where $T_{k,i}$ is the i^{th} period of the k^{th} harmonic.

Instead of the unit impulses we use impulses of different amplitudes as inputs of the system:

$$u_k = [u_{k,1}, u_{k,2}, \dots, u_{k,M}]. \qquad (7)$$

where $u_{k,i}$ is the ith input of the k^{th} harmonic. The input value $u_{k,i}$ is calculated as the maximum amplitudes of the ith period of the k^{th} harmonic. The detail procedure of determining inputs is presented in the paper by Pyž et al. (2014).

Such inputs and periods changing gives naturalness to the synthesized sound.

3 SINEWAVE MODELING BASED ON FREQUENCY TRACKERS

Our goal is to extract information from the speech signal phoneme for sinewave modeling. For this purpose, the tracking technique is used. According to this technique, the magnitude spectrum $|X_l(k)|$ of the phoneme signal $x(t)$ is calculated and the detection of local peaks in the spectra is performed. The list of estimated frequencies and amplitudes of the detected sinusoidal peaks is fed into to the tracking algorithm. In this paper, we use a classical algorithm proposed by McAulay and Quatieri (1986). According to this algorithm, the process of matching each spectrum peak in frame k to the peaks in frame k+1, is given in the three following steps:

Step 1. For each frequency ω_i^k in frame k the frequency ω_j^{k+1} in frame k+1 is sought, which is closest to such a frequency and whose absolute distance is less than the threshold Δ. This condition can be expressed by the following formula:

$$|\omega_i^k - \omega_j^{k+1}| < |\omega_i^k - \omega_p^{k+1}| < \Delta \qquad (8)$$

where $i = 1, \dots, L_k$ (L_k – the total number of peaks in frame k), $j = 1, \dots, L_{k+1}$, (L_{k+1} – the total number of peaks in frame $k+1$), and $(p = 1, \dots, L_{k+1}) \cap (p \neq j)$.

If the condition (8) is satisfied, then ω_j^{k+1} is declared to be a candidate to ω_i^k. Otherwise, if the absolute distance between all frequencies ω_j^{k+1} and frequency ω_i^k is greater or equal than the threshold Δ, i.e.:

$$|\omega_i^k - \omega_j^{k+1}| \geq \Delta \qquad (9)$$

then the frequency ω_i^k is matched to itself in a frame k+1, but with zero amplitude, and is eliminated from further consideration.

Step 2. In this step, it is checked, if ω_j^{k+1} has no better match to unmatched frequencies of frame k. This condition can be defined as follows:

$$\left|\omega_j^{k+1} - \omega_i^k\right| < \left|\omega_j^{k+1} - \omega_{p+1}^k\right| \tag{10}$$

where $p > i$.

If condition (10) is satisfied, then ω_i^k is matched to ω_j^{k+1} . Contrarily, the adjacent remaining lower frequency ω_{j-1}^{k+1} (if such exists) is tested. If the absolute distance between this frequency and the frequency ω_i^k is less than the threshold Δ, the match of these frequencies is made. Otherwise, the frequency ω_i^k is matched to itself with zero amplitude.

Step 3. For the remaining frequencies in frame k +1, for which no matches have been made frequencies are created in frame k with zero magnitude and the match is made.

Based on the obtained analysis, resynthesis is then performed. For reconstruction a sound from a track, we use a sinewave oscillator bank developed by Ellis (Ellis, 2004).

4 EVALUATION OF LOMBARD EFFECT IN MODELS

The speech signal is characterized by many features such as phonemic variation, temporal structure, prosody, voice timbre and quality (Ellis, 2008). It also contains different components of the speaker's profile like emotions or sentiments. All these features are connected into a one-dimensional signal. Therefore, observation and detection of speech changes in the context of Lombard speech are complex. Due to the occurrence of LE, the average intensity of the signal increases, frequencies of the fundamental and formants are shifted, duration of the individual words and that of vowels is changed, pause between words may be shortened, spectral energy from the low-frequency band may shift towards the medium and high-frequency bands (Folk and Schiel, 2011; Godoy et al., 2014; Garnier et al., 2006).

Based on these indications given above, we investigate a set of parameters including time and frequency domain features. It is because some of the changes in speech may be better visible in the signal descriptors rather than basic signal analysis. The time domain parameters are based on the analysis of the distribution of sound sample values in relation to zero and Root Mean Square (RMS) energy of the signal. Parameters contained in this group are as follows: the number of samples exceeding levels RMS, 2 × RMS, 3 × RMS; the number of the signal crossings in relation to levels RMS, 2 ×RMS, 3 ×RMS (Kostek et al., 2011).

The frequency domain parameters are calculated from the Discrete Fourier transform spectrum. The following spectral shape parameters are extracted based on MPEG 7 standard: Audio Spectral Centroid, Audio Spectral Spread, Audio Spectral Skewness, Audio Spectral Kurtosis, Spectral Entropy, Spectral RollOff, Spectral Brightness, Audio Spectrum Envelope, Spectral Flatness Measure. The spectrum shape parameters let us observe the change of spectrum shape in the context of noise. Due to the fact that Mel-Frequency Cepstral Coefficients (MFCCs) play major role in most applications considering speech in noise analysis (Al-Ali et al., 2017; Leu and Lin, 2017), they are also included in our parameter set. We calculated 20 MFCC coefficients. The scale of the first 13 filters is linear; for the rest of filters, the scale becomes logarithmic. The width of the linear filter is 66.67 Hz. The first four formants (F1-F4) are also included in our parameter set. The detailed description of selected parameters is given in several authors' publications (Korvel et al., 2019; Kostek et al., 2011; Rosner et al., 2014; Rosner and Kostek, 2018).

Our goal is to determine if the synthesized signal retains information concerning the Lombard effect. For this purpose, an analysis of noise residual is performed. The analysis of the speech signal noise residual based on parametric description and examination of the extracted parameters is used. The procedure consists of three steps:

Step 1: The analysis of data samples recorded in various noise environments.

Step 2: Most discriminating parameter selection.

Step 3: Parametrization of noise residual signal.

Step 4: Classification of speech samples.

In Step 1, the analysis of data samples recorded in various noise environments is performed. For this purpose, the parameters described above are calculated. Before the parameter calculation process, the speech signal is divided into short-time segments. The length of the segment is an integer power of 2, and the overlap between adjacent segments is equal to 50%. A vector of acoustic parameters is extracted for each segment. Then, the mean value is computed based on these parameters obtained from all short-term segments.

In the second step, the acoustic parameters which show differences between signals recorded with different types of noise and without them are determined. The analysis was performed for each phoneme separately.

Then, the parameterization of noise residual of all considered phonemes is performed. The parameters determined in the previous step are extracted.

The last point of the analyses (Step 4) is the classification of speech samples. For this purpose, the extracted parameters are normalized. It was decided to normalize the values to the range between 0 and 1. The classification is performed based on the parameters extracted from noise residual. In the experiment, two classical machine learning algorithms to compare classification rates are used. The first of them is k-Nearest Neighbors (*k*NN) based on the calculation of distances between parameters. The value k was set to 7. Also, a Naive Bayes classification method based on Bayes theory (Kotsiantis, 2007) is employed for comparison purposes.

5 SPEECH RECORDINGS AND DATA EXTRACTION

The recordings of sound samples were carried out in a room with an acoustically treated interior which suppresses reverberation. The recordings were made in two conditions: in the room without additional noise as well as with interference. The interference was given as pink noise generated using the noise generator, and the natural language samples of babble speech played back. As a result, four types of speech recordings were obtained. They are shown in Table 1.

Table 1: Types of Speech Recordings.

The study room with acoustic barriers
• recordings without noise • recordings with pink noise of approximately 73 dB • recordings with pink noise of approximately 84 dB • recordings with babble speech of approximately 80 dB

The experiment consists of extraction of the vowel and consonant phonemes from the recordings of utterances of three females and three males. The recording scenario included sentences and single words read in Polish. Information about audio data is given in Table 2.

The recordings have been segmented at the phoneme level. The annotation was manually conducted using PRAAT program.

Table 2: Parameters of Audio Data.

Format	wav
Sampling frequency	48 kHz
Quantification	16 bits
Number of channels	1

6 EXPERIMENT RESULTS

The results of the modeling of the speech signal with the Lombard effect are given in this Section. The experiments were performed in the MATLAB environment using tools created by authors. An example of the analyzed phoneme signals is shown in Figure 1.

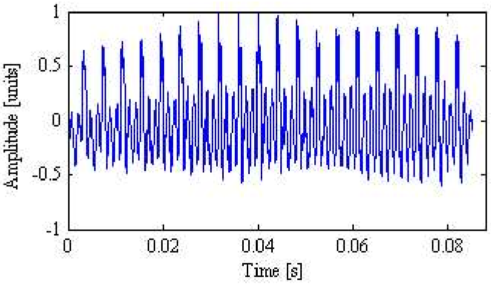


Figure 1: The Waveform of the Analyzed Speech Signal (Phoneme /a/).

An experiment begins with expanding speech signals into harmonics using rectangular filters (Eq. (3)-(4)) that are implemented by the inverse Fourier transform. Then each harmonic is modeled as the input of the SISO system. For this purpose, the parameters of the impulse response (see Eq. (2)), as well as system inputs and periods, are determined. Parameters of the 1st – 3rd component impulse responses are shown in Table 3.

Table 3: Parameters of the 1st – 3rd Component Impulse Responses of the Phoneme /a/.

Component number	1	2	3
f	241	482	723
λ	-600	-600	-600
a_1	0	0	0
a_2	2.02307	1.14826	3.15181
a_3	0.01133	0.01267	0.04995
a_4	0.00016	0.00012	0.00035
φ_1	0	0	0
φ_2	2.91723	-2.06651	-1.16040
φ_3	-0.70553	0.87301	1.59821
φ_4	3.07801	-1.56559	-0.51482

The periods of the SISO system are presented in Figure 2.

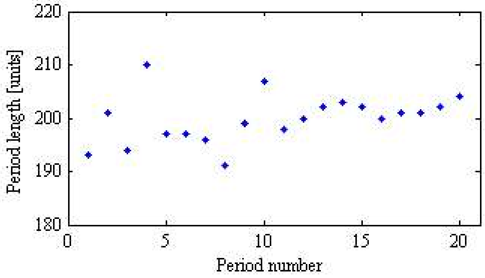


Figure 2: The Periods of the SISO System (Phoneme /a/).

In each period of the selected harmonic, we find the maximum of the amplitude. The amplitude maxima for the first three components are shown in Figure 3.

The 1st speech signal harmonic and the 1st harmonic of the SISO system are shown in Figure 4. In order to evaluate the accuracy of modeling, the spectrum of the real data and modeled harmonics sum have been compared. The mean absolute error (MAE) is utilized in the model evaluation (Chai and Draxler, 2014):

$$MAE = \frac{1}{Q}\sum_{q=1}^{Q}\left|S_q - \hat{S}_q\right| \cdot 100\% \tag{11}$$

where S_q is the q^{th} value of the spectrum of the real phoneme, and $\hat{S}_q$ is the q^{th} value of the spectrum of the modeled phoneme (Q refers to the spectrum length).

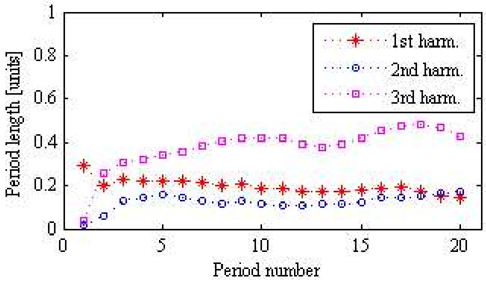


Figure 3: The Amplitude Maxima for the First Three Harmonics (Phoneme /a/).

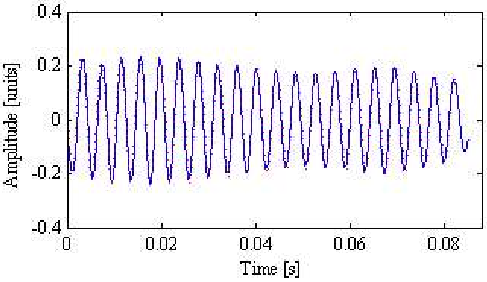


Figure 4: The 1st Speech Signal Harmonic and its Model ('+' – the Real Data, the Solid Line – the Output of the SISO System).

For the phoneme /a/, shown as an example in Figure 4, MAE is 3.62%.

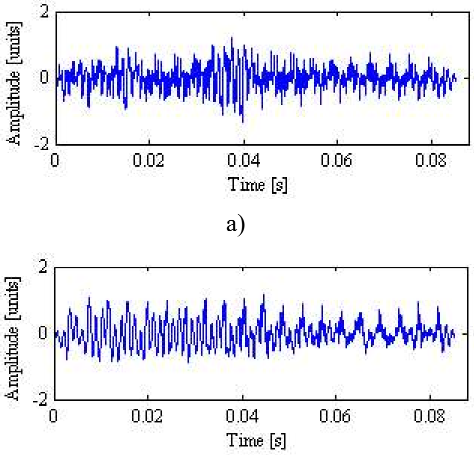
We carried out the experiment using utterances of three females and three males for vowels and semivowel phonemes modeling. The MAE values of the estimated signal spectrum depending on noise level are given in Table 4.

Table 4: The MAE for the Estimated Vowel and Semivowel Phoneme Signal Spectrum.

	Phoneme number	Without noise	Pink noise 73 dB	Pink noise 84 dB	Babble speech 80 dB
/a/	61	5.19%	5.67%	6.62%	6.22%
/i/	65	3.13%	3.69%	4.36%	4.94%
/e/	90	4.44%	5.35%	5.70%	5.37%
/o/	96	3.38%	4.19%	4.62%	4.40%
/u/	68	2.32%	2.77%	3.36%	3.16%
/l/	27	2.50%	4.13%	3.55%	2.88%
/m/	22	2.12%	3.03%	3.56%	3.10%
/n/	75	2.42%	2.90%	3.26%	2.75%
/j/	29	3.09%	4.51%	4.61%	4.26%
/r/	37	3.95%	4.30%	4.54%	4.06%
/w/	11	3.20%	2.53%	3.64%	3.39%
All phonemes	581	3.25%	3.92%	4.35%	4.05%

The data given in Table 4 also show how many recordings were used in the experiment for each phoneme.

In the second part of the experiment, a parametric analysis of the synthesized signal residual is performed. In order to evaluate the obtained results, the signal residual given from sinewave sum is used. An example of residual signals is shown in Figure 4.



a)

b)

Figure 5: The Signal Residual of a) the Harmonic Sum Generated as the Input of the SISO Model, b) Sinewaves.

An analysis was performed for each phoneme separately. Evaluation of the method suitability is based on the calculation of the acoustic parameters for the residual signal. We extracted residual parameters for all the phonemes. Classification based on parameters described in Section 4 was performed. The parameter set of each phoneme is divided into two parts: training dataset and testing dataset. For the class determination, kNN and Naive Bayes classifiers are used. The overall accuracy of the residual classification is given in Table 5, where the following denotations are used:

- A – recordings without noise and with pink noise of 73 dB;
- B – recordings without noise and with pink noise of 84 dB;
- C – recordings without noise and with babble speech of 80 dB.

In order to evaluate the classifier performance, the average accuracy was calculated. A comparison of the performance of two selected classification methods averaged for all phonemes is given in Table 6.

The lowest residual classification accuracies have been achieved in the case of the harmonic sum (Table 6). In order to determine whether the differences between the means of the residual classification for sinewave and harmonic sum are statistically significant, the t-test for two independent means was used (Lee, 2014).

Table 5: The Results of the Residual Classification.

		Real speech phoneme		Residual of a sinewave sum		Residual of a harmonic sum	
		kNN	Naive Bayes	kNN	Naive Bayes	kNN	Naive Bayes
/a/	A	76.3%	65.8%	65.8%	65.8%	76.3%	57.9%
	B	92.1%	84.2%	81.6%	84.2%	81.6%	79.0%
	C	94.7%	94.7%	92.1%	94.7%	79.0%	76.3%
/i/	A	70.0%	70.0%	62.5%	72.5%	65.0%	45.0%
	B	75.0%	77.5%	72.5%	80.0%	70.0%	75.0%
	C	72.5%	70.0%	72.5%	72.5%	57.5%	57.5%
/e/	A	94.6%	94.6%	85.7%	89.3%	80.4%	82.1%
	B	94.6%	94.6%	89.3%	91.1%	82.1%	85.7%
	C	100.0%	96.4%	98.2%	96.4%	91.1%	76.8%
/o/	A	60.3%	53.5%	56.9%	58.6%	43.1%	50.0%
	B	84.5%	84.5%	81.0%	82.8%	72.4%	79.3%
	C	67.2%	65.5%	67.2%	65.5%	41.4%	62.1%
/u/	A	66.7%	66.7%	57.1%	61.9%	61.9%	59.5%
	B	69.1%	76.2%	69.1%	69.1%	64.3%	69.1%

Table 5: The Results of the Residual Classification. (Cont.).

	C	54.8%	54.8%	52.4%	45.2%	45.2%	50.0%
/l/	A	77.8%	72.2%	72.2%	66.7%	50.0%	61.1%
	B	83.3%	77.8%	72.2%	77.8%	77.8%	66.7%
	C	77.8%	77.8%	72.2%	61.1%	44.4%	50.0%
/m/	A	78.6%	71.4%	71.4%	78.6%	71.4%	64.3%
	B	64.3%	64.3%	78.6%	71.4%	57.1%	50.0%
	C	71.4%	50.0%	50.0%	50.0%	50.0%	50.0%
/n/	A	69.6%	60.9%	63.0%	63.0%	54.4%	56.5%
	B	67.4%	56.5%	52.2%	52.2%	56.5%	56.5%
	C	67.4%	65.2%	69.6%	54.4%	54.4%	45.7%
/j/	A	88.9%	83.3%	61.1%	83.3%	72.2%	72.2%
	B	94.4%	88.9%	83.3%	83.3%	77.8%	72.2%
	C	88.9%	72.2%	83.3%	72.2%	83.3%	55.6%
/r/	A	75.0%	66.7%	50.0%	62.5%	66.7%	58.3%
	B	87.5%	83.3%	79.2%	70.8%	62.5%	66.7%
	C	75.0%	70.8%	66.7%	75.0%	50.0%	16.7%
/w/	A	100.0%	65.0%	62.5%	37.5%	62.5%	37.5%
	B	87.5%	100.0%	75.0%	87.5%	37.5%	75.0%
	C	75.0%	50.0%	62.5%	62.5%	62.5%	50.0%

Table 6: The results of the residual classification averaged for all phonemes.

	Real speech phoneme	Residual of the sinewave sum	Residual of the harmonic sum
kNN			
A	78.0%	64.4%	64.0%
B	81.8%	75.8%	67.2%
C	76.8%	71.5%	59.9%
Mean	78.9%	70.6%	63.7%
Naive Bayes			
A	70.0%	67.2%	58.6%
B	80.7%	77.3%	70.5%
C	69.8%	68.1%	53.7%
Mean	73.5%	70.9%	60.9%

The test significance level equals 0.05. These results are as follows: for kNN classifier, the t-value is 2.11605, the *p*-value is 0.019118, for Naive Bayes classifier the t-value is 2.80706, the *p*-value is 0.003309. Therefore, we can conclude that the differences are significant.

7 CONCLUSIONS

Phoneme models based on SISO system are proposed in this paper to create a synthesized speech that retains the Lombard effect. For that purpose, sounds of vowels (/a/, /i/, /o/, /u/, /e/) and semivowels (/l/, /m/, /n/, /j/, /r/, /w/) were utilized in this research. In general, 581 phoneme recordings employed in the experiment construction.

In the first part of the experiment, models of normal speech and that with Lombard effect were created. The average modeling accuracies (MAE of the modeled and real signal spectrum) resulted from this part are as follows: 3.25% for recordings without noise, 3.92% for recordings with pink noise of 73 dB, 4.35% for recordings with pink noise of 84 dB, and 4.05% for recordings with babble speech of 80 dB.

In the second part of the experiment detection of the Lombard effect in the synthesized signal noise residual was performed. We observed that the lowest residual classification accuracies (63.7% for kNN classifier, and 60.9% for Naive Bayes classifier) were obtained in the case of harmonic sum-based synthesis. In the case of the sinewave sum, the mean classification accuracy for the kNN classifier was 70.6%, while for the Naive Bayes classifier - 70.9%. The employment of the one-way analysis of means test (t-Test) revealed that the differences between accuracies of the two considered synthesis methods are significant.

The results obtained in this study lead to the conclusion that the proposed model may retain Lombard effect characteristics.

In the future, we would like to pursue the analysis of the synthesized phonemes in the context of checking whether the models created are language-dependent.

Moreover, future research will expand the database so that it can be possible to compare the results obtained with the state-of-the-art algorithms, such as neural networks (and specifically convolutional neural networks). The authors have experience in such an analysis (Korvel et al., 2018), but even though it will not be possible to directly compare the results, because, in the case of deep learning, 2D signal representations will be used (cepstrogram, spectrogram, etc.).

Additionally, in the case of speech synthesis, an essential element is the subjective test that allows for assessing the quality of the synthesized sounds obtained. This aspect is especially interesting in the context of language specifics. Preliminary, informal tests show that quality of the synthesized phonemes may be directly compared to the original sound. Therefore, the subjective quality evaluation will be based on formal listening test sessions in which normal-hearing subjects will participate. The original phoneme, as well as the corresponding synthesized versions, will be used. Subjects will be asked to answer the following question: "Does the phoneme sound natural?" and to assign a corresponding score. Then, the participants will have to distinguish between the original phoneme and the synthesized one in the AA-AB comparison test, where A is the original sound and B the synthesized phoneme. Thus, this will be thoroughly researched in the future.

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