Hands-Free Wearable Electrolarynx using Linear Predictive Coding Residual Waves and Listening Evaluation

Masaki TAKEUCHI,^{*,#} Jaesol AHN,^{**} Kunhak LEE,^{***} Ken TAKAKI,^{*} Tohru IFUKUBE,[†] Ken-ichiro YABU,[†] Shinnosuke TAKAMICHI,^{††} Rumi UEHA,^{†††} Masaki SEKINO[‡]

Abstract A conventional electrolarynx (EL), which is used by laryngectomees, produces monotonous sound and occupies a user's hand; hence, we developed a hands-free wearable device that improves voice quality. The proposed device estimates individual vocal tract features using linear predictive coding (LPC) and generates sound vibrations using an LPC inverse filter. Additionally, we reproduced the vibration sound using a transducer and amplified the first harmonic frequency and the second one. We conducted an objective experiment to compare the spectra of natural voice, a conventional EL, and the proposed device. We also conducted a subjective experiment in which we asked healthy subjects to listen to and evaluate the conventional EL and the proposed device. The results of the objective experiment demonstrated that our model was characterized by two formant peaks that were similar to the conventional EL and the natural voice. The results of the subjective experiment demonstrated that our model was more powerful and clearer than the conventional EL. These findings indicate that the voice of our device is spectrally close to human voice and gives the audience a more powerful and clearer sound.

Keywords: welfare device, signal processing, transducer.

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^{*} Graduate School of Engineering, The University of Tokyo, Tokyo, Japan.

^{**} Faculty of Information and Communication Engineering, The University of Tokyo, Tokyo, Japan.

^{***} Faculty of Mechanical Engineering, The University of Tokyo, Tokyo, Japan.

[†] Research Center for Advanced Science and Technology, The University of Tokyo, Tokyo, Japan.

^{††} Department of Information Science and Technology, The University of Tokyo, Tokyo, Japan.

^{†††} Swallowing Center, The University of Tokyo Hospital, The University of Tokyo, Tokyo, Japan.

[‡] Department of Bioengineering, The University of Tokyo, Tokyo, Japan.

[#] 2-11-16 Yayoi, Bunkyo-ku, Tokyo 113-0032, Japan.

E-mail: takeuchi-masaki854@g.ecc.u-tokyo.ac.jp

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1. Introduction

People typically communicate using their voices. The human voice is produced as follows. First, exhaled air flows from the lungs to the larynx. Thereafter, the mucous membrane of the vocal cords repeatedly opens and closes at high speed, further causing the airflow from the lungs to vibrate and produce sound. Finally, owing to the differences in the shapes of the mouth and tongue, the resonance frequency in the vocal tract varies, further creating different types of sound, thus producing the voice. However, people whose vocal cords have been removed because of laryngeal cancer cannot send air to the vocal cords, and consequently are not capable of producing sound. Hence, they have speech impediments, because although they can move their mouth, they are unable to produce sound.

One way by which such people can regain their voice is by using an electrolarynx (EL). An EL user presses a button on a cylindrical machine to generate electrical vibration. However, the user cannot speak unless he/she holds the EL with one hand and presses it against the throat. Vibrations of only a single frequency can be generated, and the sound will be monotonous and mechanical, regardless of the user.

A few researchers have attempted to address these

challenges by developing a hands-free EL. Uemi et al. [1] developed a hands-free pitch-control EL based on the expiration pressure. Hashiba et al. [2, 3] developed and evaluated a wearable EL equipped with a wireless on/off switch and pocket-sized controller. Although a hands-free model could be realized, its voice quality and usability, which are important for users, were mediocre. Matsui et al. [4] developed a hands-free EL using a micro-electromechanical system accelerometer to control the pitch via arm movement. Two types of pitch contour generation methods were proposed and tested along with a conventional EL. However, the vibration sound was similar to a robotic sound, and people without the ability to control their arms could not use it.

Other hands-free ELs include electromyography (EMG) sensors. Goldstein et al. [5] first designed and implemented a hands-free EL that could be controlled using EMG. Subsequently, some researchers developed a hands-free EL using neck and face surface EMG (sEMG) to control voice initiation, voice termination, and pitch [6–9]. However, these studies did not realize a natural voice.

Moreover, several studies have been conducted on improving the sound quality using machine learning [10–15], noise reduction [16–20], or both [21]. Although these systems improve the intelligibility of the output voice, the speaker outputs the voice. Thus, users cannot generate syllables from their mouths. This causes a delay between the movement of the mouth and the generation of voice.

Accordingly, this study aims to develop a novel hands-free EL (**Fig. 1**) for voice restoration with sound vibrations similar to those of the human voice. Moreover, we evaluated the output voice of our device by spectral comparison and voice listening experiments.

Fig. 1 Photograph of a person using the proposed device.

2. Materials and Methods

2.1 Device structure

Figure 2 depicts the proposed system. The sound vibrations, which are generated from a computer or smartphone, are sent to transducers via Bluetooth and a cable. The proposed device had two cylindrical transducers that were placed on the throat while speaking. The device vibrated the throat to produce natural sounds through the vocal tract. To generate a natural sound, we estimated sound vibrations from raw voice in advance, as described in **Section 2.2**. We also used an EL transducer and a sound actuator. The EL transducer was a hitting-type transducer, and was used in a conventional EL. The sound actuator was the ACOUSTICHAPTIC® Actuator, LA-0031, $\varnothing 20 \times 30$, provided by Foster Electric Co., Ltd. We used two different types of transducers to achieve a wider frequency range and louder volume. The EL transducer had a narrower frequency range of 60–200 Hz, but its output volume was louder. However, the output volume of the sound actuator was lower than that of the EL transducer. Two different types of vibrators were needed to produce sounds for both men and women of all ages, as the EL transducer had a narrower frequency range but greater vibration, while the sound actuator had a wider frequency range but smaller vibration.

2.2 Generation of natural sound vibrations

We used linear predictive coding (LPC) to estimate the sound vibrations. LPC assists in extracting parameters of vocal tract features [22]. LPC residue is one of the major driving signals in source-filtered speech synthesis. In fact, some studies have demonstrated that it can produce vowels in a manner similar to that of the human voice when applied to the EL [23]. To acquire the prediction function for minimizing the difference between the predicted and actual values, a linear combination of values obtained from previous samples was used as the predict-

Fig. 2 System design.

ed value, which is denoted by x_n , and is expressed as follows:

$$x_n = -(a_1 s_{n-1} + a_2 s_{n-2} + \dots + a_p s_{n-p}), \quad (1)$$

where a_n is the linear predictor coefficient, and p is the LPC order. s_n is the actual recorded data at time n , and e_n is the prediction error within the sampling interval. a_n was calculated to minimize the squared value of e_n , which is expressed as

$$e_n = s_n + a_1 s_{n-1} + a_2 s_{n-2} + \dots + a_p s_{n-p}. \quad (2)$$

Thereafter, Eq. (2) was z -transformed as follows:

$$\begin{aligned} E(z) &= S(z) \left(1 + a_1 z^{-1} + \dots + a_p z^{-p} \right), \\ S(z) &= \frac{1}{1 + a_1 z^{-1} + \dots + a_p z^{-p}} E(z), \\ S(z) &= H(z) E(z) \left(\because H(z) = \frac{1}{1 + a_1 z^{-1} + \dots + a_p z^{-p}} \right), \end{aligned} \quad (3)$$

$$E(z) = \frac{1}{H(z)} S(z). \quad (4)$$

$\frac{1}{H(z)}$ indicates an LPC inverse filter. The vocal tract parameters can be removed by the human voice by multiplying the LPC inverse filter. This results in LPC residual waves. We recorded two voices to estimate the actual LPC residual wave: a voice generated with the transducer and a human voice with normal tone. A chirp signal, which is typically used to measure the frequency characteristics, was used for speaking with the transducer. The user opened the mouth in a manner that pronounced the Japanese vowel /a/ and placed the transducer that generated the chirp signal on the throat. The recording duration was 5 s. Let $X(z)$ and $Y(z)$ be the z -transforms of the chirp signal and the signal output from the mouth, respectively, then $H(z)$ can be expressed as

$$H(z) = \frac{Y(z)}{X(z)}. \quad (5)$$

To record a normal tone, the user said /a/ using the vocal cord. Let $Y'(z)$ be the z -transforms of the output voice and LPC residual wave, then $X'(z)$ can be expressed as

$$X'(z) = \frac{Y'(z)}{H(z)}. \quad (6)$$

Subsequently, we calculated the LPC residual wave, as shown in **Fig. 3**. Unlike the waves generated by the human voice, the sound pressure levels of the first harmonic frequency and the second harmonic are less than approximately 20 dB (**Fig. 3b**).

Therefore, using an audio equalizer in Audacity, we increased the levels of the first harmonic frequency and second harmonic in the output of our device to obtain amplitudes similar to the human voice waves. The first harmonic frequency is the smallest periodic interval of a periodic speech signal. If the voice waveform is enlarged,

Fig. 3 Frequency spectrum of (a) a voice generated with LPC and (b) of a human voice.

the same waveform is repeated constantly. When repeated every 0.01 s, it is repeated 100 times in 1 s. In other words, the first harmonic frequency is 100 Hz. The first harmonic frequency is also known as F0. In this case, the first harmonic frequency is 100 Hz. The second harmonic is twice the value of the first harmonic frequency.

2.3 Spectrum comparison

We compared the frequency spectra of our device, the conventional EL, and natural voice, without any change in the first harmonic frequency (F0). This is the pitch of the voice. All voices were produced by the lead author of this paper. He is not a laryngectomee. However, there is no significant difference in the shape of the vocal tract between laryngectomees and non-laryngectomees. Hence, no difference in the test results is expected. The recording environment comprised a PC (Panasonic, denoted as CF-5V hereinafter), a microphone (Audiotechnica, ATR1100x), a USB audio conversion adapter (Planex, PL-US35AP), and Praat, which is the application software used to record and conduct the listening evaluation experiments [24]. The recorded phrase was “Konnichiwa,” which means “good afternoon” in Japanese. We extracted the vowel “a” for 0.06 s and Fourier-

transformed the signals to display the frequency spectrum.

2.4 Listening evaluation

Eight subjects aged between 20 and 40 years with normal hearing participated in this study. They listened to the voice as two different types of speech (generated with a conventional EL and with the proposed device) and a total of 20 ATR sentences, ten each for the conventional EL and the proposed device. The A-weighted filtering sound levels were unified using MATLAB. A-weighted filtering is a sound pressure level used as a noise loudness measure, similar to human hearing [25]. We used the Praat Experiment MFC to conduct a multiple forced choice listening experiment for the listening evaluation. The order of the listening sounds was randomized. Our experiment used the SD method, which was used in previous studies as a subjective method to evaluate the substitute vocalization of laryngectomized patients [26, 27]. The SD method is a 7-point grading method for two adjectives A and B with polar meanings as follows: 1) very likely A; 2) likely A; 3) fairly likely A; 4) neither likely A nor likely B; 5) fairly likely B; 6) likely B; and 7) very likely B. The scales used were selected from a previous study [27], but we extracted the voice quality scales only, and eliminated scales for tempo and pitch. There was no breathing involved in this vocalization, therefore “breathy - not breathy” was excluded. We also excluded “slack - firm,” which is not a good scale for comparison of voice quality. Then, we added scales related to naturalness [28], individuality [29], and listening effort [30], which are subjective scales commonly used in the speech synthesis field and research on substitute vocalization in laryngectomized patients. Thus, the list of scales is as shown in **Table 1**.

In this experiment, the lead author of this paper read ATR sentences using the conventional EL and the proposed device. ATR sentences are phonetically balanced

sentences extracted from newspapers, journals, novels, letters, and textbooks, among others, such that several phonetic environments occur at the same rate as much as possible [31]. We used 20 ATR sentences from A31–A50. Two experimental sets were created and designated A and B. Set A included 10 sentences (A31–A40) spoken with the conventional EL and 10 sentences (A41–A50) spoken with the proposed device. Set B was vice versa. All eight subjects participated in the experiment, four in each set. The experimental data were analyzed by Shapiro-Wilk test. As the experimental data were not normally distributed, Mann-Whitney U test was used to determine whether there was a significant difference between the conventional EL and the proposed device for each scale, using R version 4.0.3. Statistical significance was set at $p < 0.05$. We divided the experiments into practice and test sessions. All procedures performed in this study were in accordance with the ethical standards of the Ethical Review Committee of the University of Tokyo. All participants provided written informed consent.

3. Results and Discussion

To investigate the formants of the methods, we measured their spectra. A formant is a peak in the spectral envelope produced by the resonance of the vocal tract. **Figure 4** shows the spectra of the natural voice, the EL output voice, and the output voice produced by the proposed device, respectively. **Figure 4a** shows the peaks at the first formant, F1 (790 Hz), and at the second formant, F2 (1,190 Hz). **Fig. 4b and 4c** show that the EL and the proposed device output specify peaks at F1 (790 Hz) and F2 (1,190 Hz), which represent the natural voice.

As depicted in **Fig. 4c**, our device reproduced the third formant (F3) at 2,700 Hz, and the fourth formant (F4) at approximately 3,500 Hz. F3 and F4 are related to personality and sound quality, respectively [22]. However, the proposed device did not reproduce the fifth formant (F5) at approximately 6,000 Hz. As depicted in **Figs. 4b and c**, the frequency spectra of the EL and the proposed device did not show a clear formant after 4,000 Hz. Conversely, **Fig. 4a** shows that natural speech had a clear formant after 4,000 Hz. This is due to mechanical impedance mismatch, which causes reflection from the oscillator to the oral cavity. The mechanical impedance of the skin present in the oral cavity of humans acts as a low-pass filter, which filters sound sources of 4,000 Hz and above [32]. This is presumably why F5 at 6,000 Hz could not be reproduced.

Figure 5 depicts the evaluation results of the conventional EL and the proposed device for each scale. These results suggest that there were significant differences with respect to “beautiful” ($p < 0.05$), “clear” ($p <$

Table 1 List of scales used in this study.

number	negative	positive
1	ugly	beautiful
2	not intelligible	intelligible
3	weak	powerful
4	expressionless	expressive
5	melodious	monotonous
6	dull	clear
7	unnatural	natural
8	not individual	individual
9	bad listening effort	good listening effort

Fig. 4 (a) Natural voice, (b) EL output voice, and (c) proposed device voice for the Japanese vowel “a”; LPC spectral envelope of 0–6,000 Hz.

Fig. 5 Evaluation results of the experiment.

0.05) and “powerful” ($p < 0.001$). For the “beautiful” scale, the conventional EL performs better than the proposed device. For the “clear” and “powerful” scales, the proposed device performs better than the conventional EL.

Significantly, the sound produced by our device was not more beautiful than that of the conventional EL. One participant claimed that he heard noises similar to the sound of a buzzer. The vibration sound from the transducer could not move efficiently through the neck surface. The reflected sound was the noise of the vibration sound from the transducer. This noise is the main reason for the difficulty in generating a beautiful output voice using the proposed device.

In future work, we will attempt to attenuate the vibration noise from the transducer. Moreover, a method to improve the vibration sound using an LPC residual wave was designed specifically for the lead author’s voice. The validity of the proposed device must also be examined in other people.

4. Conclusion

In this study, we developed a device that assists laryngectomees to speak from the mouth, aiming to achieve sound vibrations similar to those of the human voice. We developed a novel hands-free device and generated sound vibrations using LPC residual waves. We conducted an objective spectral evaluation, which demonstrated that the proposed device produced a voice with characteristics similar to those of the natural voice and the conventional EL at 0–2,000 Hz. Furthermore, F3 and F4, which mainly reflect individuality, were superimposed on the artificial laryngeal voice. We verified the possibility of evaluating nine bipolar scales using the SD method. The results of listening evaluation demonstrated that the proposed device was more powerful and clearer than the conventional EL. Conversely, the sound produced by the conventional EL was more beautiful than that produced by the proposed device.

Conflicts of interests

The authors declare no conflicts of interest with any companies or commercial organizations according to the definition of the Japanese Society for Medical and Biological Engineering.

Ethics declaration

This study was conducted in accordance with the ethical principles of human subject research, and informed consent was obtained from each subject. The study was approved by the University-wide Ethics Review Committee of the University of Tokyo. The authorization number is KE21–70.

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Masaki TAKEUCHI

Masaki TAKEUCHI received his master's degree in Engineering from the University of Tokyo, Japan, in 2021. Since 2021, he has been pursuing his doctoral degree in the Graduate school of Engineering at The University of Tokyo.

Jaesol AHN

Jaesol AHN graduated from Changwon Science High School in 2015. He is currently an undergraduate student in the Department of Electronic Information Engineering, Faculty of Engineering, the University of Tokyo.

Kunhak LEE

Kunhak LEE graduated from Yuseong High School in 2017. He is currently an undergraduate student in the Department of Mechanical Engineering, Faculty of Engineering, the University of Tokyo. His research interest includes Microfluidics. He is going to be a member of HTSJ (The Heat Transfer Society of Japan) in April 2022.

Ken TAKAKI

Ken TAKAKI received his master's degree in Engineering from the University of Tokyo, Japan, in 2021. Since 2021, he has been pursuing his doctoral degree in the Graduate school of Engineering at The University of Tokyo.

Tohru IFUKUBE

Tohru IFUKUBE received his master's degree from Hokkaido University, Japan, in 1971. From 1971 to 1989, he worked as an assistant professor at the Research Institute of Applied Electricity, Hokkaido University, Japan. He was a visiting associate professor at Stanford University, Stanford, USA, in 1984. From 1989 to 2002, he worked as a Professor at the Research Institute of Applied Electricity, Hokkaido University, Japan. From 2002 to 2007, he worked as a professor at the Research Center for Advanced Science and Technology (RCAST) at the University of Tokyo, Japan. From 2007 to 2009, he worked as an emeritus professor at Hokkaido University, Japan. From 2009 to 2011, he was an emeritus professor at the University of Tokyo, Japan. From 2011 to 2019, he was a Research Fellow at the Institute of Gerontology of the University of Tokyo, Japan. From 2015 to 2019, he worked as a professor at the Hokkaido University of Science, Japan. He has been an adviser at the RCAST, the University of Tokyo, Japan. His current research interests include assistive technology that supports the disabled and the elderly. He constructed a research methodology that circulates the basic science of psychophysiology and applied technologies, such as virtual reality and robots. He has led research on assistive devices for practical use.

Ken-ichiro YABU

Ken-ichiro YABU received his Ph.D.(Engineering) degree from Tokyo Metropolitan University in 2010. From 2010 to 2014, he worked as a project researcher at the Research Center for Advanced Science and Technology, the University of Tokyo, Japan. Since 2014, he has been working as a project researcher at the Institute of Gerontology, the University of Tokyo. And since 2020, he has also been working as a project researcher at the Research Center for Advanced Science and Technology, the University of Tokyo. His research focuses on assistive device technology based on speech signal processing and information technologies and mechatronics systems for people with disabilities or the elderly. He is a member of the International Speech Communication Association (ISCA), Acoustical Society of Japan(ASJ), the Virtual Reality Society of Japan (VRSJ) etc.

Shinnosuke TAKAMICHI

Shinnosuke TAKAMICHI received his B.E. degree from Nagaoka University of Technology, Nagaoka, Japan, in 2011, and his M.E. and Ph.D. degrees from the Graduate School of Information Science, Nara Institute of Science and Technology, Ikoma, Japan, in 2013 and 2016, respectively. Since 2018, he has been working as an assistant professor at the University of Tokyo. He has received more than 20 papers/achievement awards, including the 2020 IEEE Signal Processing Society Young Author Best Paper Award.

Rumi UEHA

Rumi UEHA graduated from Nara Medical University, Faculty of Medicine, Japan, in 2003. After working as a resident at the University of Tokyo Hospital, she worked as an otolaryngologist at NTT East Kanto Hospital, Tokyo Metropolitan Neurological Hospital, and Kameda General Hospital. From 2010 to 2012, she worked as a specially appointed clinician at the University of Tokyo Hospital, Japan. She was particularly interested in stroke, neurological intractable diseases, and dysphagia in patients with severe mental and physical disabilities. From 2012 to 2019, she worked as an assistant professor at the Department of Otorhinolaryngology, Head and Neck Surgery, University of Tokyo, Japan. She studied at the University of Michigan, USA, in 2012 and at the University of California, Davis, USA, in 2018. From 2019 to 2021, she was a project lecturer at the Department of Otorhinolaryngology, Head and Neck Surgery, University of Tokyo, Japan. Since 2021, she has been a director and associate professor at the Swallowing Center, University of Tokyo Hospital, Japan. Her specialization fields include swallowing disorders and olfactory impairment. She had undergone numerous surgeries and basic research. She is a member of the Association for Research in Otolaryngology.

Masaki SEKINO

Masaki SEKINO received his doctoral degree in Engineering from the University of Tokyo, Japan, in 2005. From 2005 to 2006, he worked as a project assistant at the Graduate School of Medicine, University of Tokyo, Japan. From 2006 to 2007, he worked as an assistant at the Graduate School of Frontier Sciences, University of Tokyo, Japan. From 2007 to 2010, he worked as an assistant professor at the Graduate School of Frontier Sciences, University of Tokyo, Japan. He was a French Government Scholarship student conducting research at the NeuroSpin Laboratory, CEA, France, in 2008. In 2009, he was an adjunct faculty member at the Bioimaging Center, Graduate School of Frontier Sciences, University of Tokyo, Japan. From 2010 to 2011, he worked as a lecturer at the Department of Electrical Engineering and Information Systems, University of Tokyo, Japan. From 2011 to 2020, he worked as an associate professor at the Department of Electrical Engineering and Information Systems, University of Tokyo, Japan. Since 2021, he has been working as a professor at the Department of Bioengineering, University of Tokyo, Japan. His current research interests include electromagnetic imaging of biological bodies and brain stimulation. He is a member of the Institute of Electrical and Electronics Engineers (IEEE), Bioelectromagnetics Society (BEMS), and International Society for Magnetic Resonance in Medicine (ISMRM).