Development of a Real-time Bionic Voice Generation System based on Statistical Excitation Prediction

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ABSTRACT

Despite the emergent progress in many fields of bionics, larynx amputees still lack a functional Bionic Voice source to overcome their voice disability. We have established the Pneumatic Bionic Voice (PBV) as a promising technology to generate a voice for larynx amputees. This summary provides an overview of our efforts to implement the PBV system in real-time together with a demonstration of its results.

The Pneumatic Bionic Voice prosthesis is an electronic adaptation of an old school mechanical voice source called the Pneumatic Artificial Larynx (PAL). The PAL is a non-invasive voice prosthesis, with an exceptionally high-quality voice, driven exclusively by respiration. This work proposes an approach to statistically model the PAL's voice generation mechanism from the respiration and to implement it in a real-time PBV system that larynx amputees can use to generate voice in continuous speech.

Author Keywords

Pneumatic Bionic Voice; Voice Prosthesis; Pneumatic Artificial Larynx; PAL; Voice conversion; Laryngectomy.

CSS Concepts

• Human-centered computing~Accessibility; Accessibility Technologies;

INTRODUCTION

Larynx amputees (laryngectomees) lose their larynx in advanced stages of throat cancer. Losing the larynx leads to the loss of vocal folds and the voice as the excitation source of speech. Voice prostheses aim to regenerate this excitation source. Similar to the missing vocal folds, larynx amputees shape this artificial excitation source into speech by moving their face and lips muscles. Among many voice prostheses, the Pneumatic Artificial Larynx still stands out, as the only non-invasive voice prosthesis with an exceptionally high voice quality [1–5]. It is interesting that the PAL is driven

exclusively by respiration to generate such high-quality voice. This has encouraged us to estimate the voice generation function of the PAL and employ it in a Pneumatic Bionic Voice source as its electronic adaptation [6].

The design approach of a PBV is to mimic the functionality of the PAL both in terms of controlling voice generation and reaching a similar degree of naturalness. We have already captured the voice onset/offset control of the PAL with 98.4% accuracy. This enabled us to develop the first real-time Bionic Voice source which a laryngectomee used to control using his respiration to speak in slow/fast speech rates [6]. To improve the PBV's naturalness, we developed a statistical respiration to voice conversion system to generate a voice close to the PAL source waveform from respiration. Using this framework in our design, substantially improved the Bionic Voice quality in our offline simulations and was verified through objective and subjective listening tests [7].

This paper takes the next step and develops this statistical Pneumatic Bionic Voice source prediction framework in real-time. We are excited to demonstrate this system in ASSETS 2019 and to proceed with its upcoming trials with the laryngectomees in the near future.

REAL-TIME PBV SOURCE GENERATION SYSTEM

System Overview

When a laryngectomee loses their larynx, an opening in the neck (stoma) is generated for them to breathe. The PAL is placed externally between this stoma and the mouth and similar to a whistle is driven by the variations of the pressure at its two ends, $p_m(t)$, $p_s(t)$ (Figure 1). In that sense, the PAL acts as a mechanical model of the human larynx with a fixed pair of vocal folds, driven exclusively by respiration to generate the excitation source waveform of speech: $e(t)$.

Figure 1. Respiration drive of the PAL source as a laryngectomee participant uses the PAL to speak with.

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To reach a similar high quality, the PBV system has to estimate, $e(t)$, the source excitation waveform of the PAL, from the underlying respiration signals. For this, we adopt the general statistical voice conversion framework by Toda et al. [8] and modify it to a respiration to voice conversion system [6]. The PAL's driving pressure $p_m(t)$, $p_s(t)$, are the inputs and the PAL's generated voice waveform, $e(t)$, is the intended output of this conversion system.

Statistical Source Excitation Prediction

Figure 2 summarizes the structure of the real-time PBV system. To estimate the source, $e(t)$ is broken into three components of: spectral envelope, f_0 and aperiodicity, extracted with WORLD vocoder [9]. The spectral envelope is further parameterized into 24 mel-cepstral coefficients. A set of 14 features are extracted from the respiration signal [10]. In the training phase, three Gaussian Mixture Models (GMMs) are trained for mel-cepstrum, aperiodicity and f_0 component estimation using joint probability density of respiration/voice features trajectory. The trained GMMs are used in the test phase and maximum likelihood parameter generation (MLPG) [11] of a trajectory of estimated values is used to achieve the final f_0 , mel-cepstrum and the aperiodicity values for each frame. The estimated PAL source waveform ($\hat{e}(t)$) is then synthesized by combining the three estimated components using a vocoder [9].

The real-time PBV system follows the same concept as the offline PBV [7] except for one main difference. In the offline PBV system, a batch-type version of the maximum likelihood parameter generation (MLPG) is used [11]. So, in the conversion stage of the system, the spectral parameters of multiple frames are generated simultaneously to benefit from feature correlations between frames. This is not feasible in the real-time system where a frame-by-frame low-delay MLPG should be employed [12]. This low-delay MLPG approach uses a recursive parameter generation algorithm already proven feasible in speech synthesis [13].

Implementation Details

The real-time PBV system receives its respiration input signals ($p_m(t)$, $p_s(t)$) at 1 kHz through a UDP port of an Arduino™ based pressure sensing system [6]. The UDP (also used in VoIP applications) is used due to its connectionless nature to eliminate any waiting time for the respiration signal to arrive. The respiration input and voice output of the system are windowed at overlapping 10ms and 25ms frames respectively, both with 5ms of shift. Every 5ms, the PBV system receives one frame (10ms) of respiration and generates 5ms of audio (voice) and sends it to the sound card.

The feature sequence in the real-time system is shaped similar to the offline PBV system with the analysis frame in the middle (n frames ahead and n frames before the analysis frame). This leads to n frames of look-ahead delay ($n * 5ms$ of frame-shift) in the real-time PBV system (in addition to hardware delays). With $n=2$ frames in this implementation the real-time estimation will have a delay of more than 10ms.

Figure 2. Real-time respiration to voice conversion system.

The system was implemented in C++ on an iMac computer with 32GB of RAM and a 4.2 GHz Intel Core i7 processor. To maintain a real-time performance, the system was synchronized with an independent external clock. For this, the system was implemented using two threads: a) one thread to perform the respiration to voice conversion, i.e. reading each respiration frame, estimating the voice and placing the synthesized audio frame in a certain memory address, b) an independent audio callback thread run by the OS to take the synthesized frame from that memory address and operate the sound card to play the synthesized sound frame.

EVALUATION

The average time delay between each frame of respiration input and the synthesized audio output for that frame was measured to be 23ms (sum of look-ahead and hardware delays). This led to a straightforward real-time performance in synthesizing a glitch-free audio by the soundcard.

The objective comparison of the real-time and offline systems showed a close match. The sum of the difference between the respiration feature extraction methods of real-time and offline PBV systems for respiration recordings of 45 seconds was in the order of 10^{-7} . The estimation accuracy of f_0 was 98% in real-time compared to the offline PBV and the two system's Mel-Cepstral distortion of the estimated spectrum coefficients was 3.9 dB. To evaluate the system as a whole, the estimated PAL excitation source waveforms were used to re-synthesize a copy of their corresponding PAL speech signal in an approach explained in our previous work [6]. The simulated speech quality of the offline and real-time PBV systems will be provided as part of the demonstration.

CONCLUSION

We implemented the first real-time Pneumatic Bionic Voice source prediction system and will demonstrate its performance in comparison with the reference PAL device. Our next step is to proceed with live trials of this system with larynx amputees.

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