PHYSICALLY INSPIRED SIGNAL MODEL FOR HARMONIUM SOUND SYNTHESIS

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ABSTRACT

The hand harmonium is arguably the most popular instrument for vocal accompaniment in Hindustani music today. However, it lacks microtonality and the ability to produce controlled pitch glides, which are both important in Hindustani music. A harmonium sound synthesis model with a source-filter structure was previously presented by the authors in which the harmonium reed sound is synthesized using a physical model and the effect of the wooden enclosure is applied by a filter estimated from a recorded note. In this paper, we propose a simplified and perceptually informed signal model capable of real time synthesis with timbre control. In the signal model, the source is constructed as a band-limited waveform matching the spectral characteristics of the source signal in the physical model. Simplifications are suggested to parametrize the filter on the basis of prominent peaks in the filter frequency response. The signal model is implemented as a Pure Data [1] patch for live performance using a standard MIDI keyboard.

1. INTRODUCTION

While Asian free reed instruments such as the sheng, the sho and the khaen have existed for over 2000 years, the development of Western musical instruments using free reeds (e.g. harmonium, accordion, harmonica, etc.) did not occur until the end of 19th century. The harmonium, also known as the reed organ or pump organ, reached the peak of its popularity in the Western world during the latter half of the 20th century. Dwarkanath Ghose is generally credited to have modified the European harmonium to invent the Indian hand-bellowed harmonium (Fig. 1) in 1884 [2]. Although the use of the harmonium in Western music declined, the hand harmonium grew in popularity to become the instrument of choice for vocal accompaniment in North Indian (Hindustani) classical music today.

However, the use of the harmonium in Indian music has attracted a lot of criticism from musicologists [3]. Due to its standard keyboard design, only a discrete set of twelve notes per octave, typically spaced at intervals of one semitone can be played on a harmonium. This is inconsistent with the non-equal temperament system in Hindustani music that uses 22 notes (referred to as shrutis) in one octave. Additionally, the harmonium cannot produce continuous pitch glides from one note to another as would be possible in a string instrument like a violin. Such a lack of capability makes the harmonium unsuitable for producing essential ornaments in Hindustani music such as meend, andolan, and gamaka, which are elements similar to glissando, portamento, and vibrato in Western music.

Figure 1: *An Indian hand harmonium (top view).*

To address these limitations in a real harmonium, the authors recently proposed a physics-based synthesis system for harmonium sounds [4]. The proposed system used a source-filter structure in which a 1D physics-based model of a free reed interacting with the air flow acted as the source and the effect of the wooden enclosure of the instrument was approximated by an all-pole filter whose coefficients were estimated from a recorded harmonium sound.

While the proposed system produced convincing results, we observed some limitations that hindered its use for real-time synthesis in a live-performance context. Consequently, a simplified signal model that perceptually replicated the physics-based synthesis model was developed.

Section 2 provides a short literature review on free reeds while in section 3, we describe in brief the physics-based system for harmonium sound synthesis from our previous work. Section 4 defines the main requirements for an equivalent signal model and describes our implementation of the signal model in Pure Data.

2. RELATED WORK

A comprehensive review of prominent acoustic studies on free reed instruments has been provided by Cottingham [5]. Early acoustic studies on free reeds have focused on understanding how self-sustained oscillations are produced in a free reed. A series of experimental and theoretical studies by St. Hilaire [6, 7, 8] suggests that the inertial effect of the upstream air flow is responsible in setting up self-oscillations in free reeds. Experimental studies by Tarnopolsky [9, 10] and Ricot [11] agree with this theory. Millot and Baumann [12] have described a minimal 1D model and a numerical solution scheme to simulate the sound from any free reed instrument. In previous work [4], we have proposed adaptations to the Millot-Baumann model to suitably match the physical setup and sound timbre of a Indian hand-harmonium.

3. SOURCE-FILTER BASED PHYSICAL MODEL OF A HAND-HARMONIUM

In the source-filter model used for the synthesis of voiced vowel sounds, a periodic glottal excitation signal determines the fundamental pitch (f0-frequency) while a linear filter representing the vocal tract imparts the timbral characteristics to the sound [13]. The source-filter model can be used effectively in situations where there is only a weak coupling between the source excitation and the resonator [14]. This is the case for the harmonium sound production. Since the metallic harmonium reeds are lightly damped, their frequency response is concentrated in a narrow band around their resonance frequencies. The effect of the wooden enclosure is hence analogous to the vocal tract. The use of a source-filter model to synthesize harmonium sounds is thus justified. The source and filter models that we used are described in the following subsections.

3.1. Physical model of the harmonium free reed

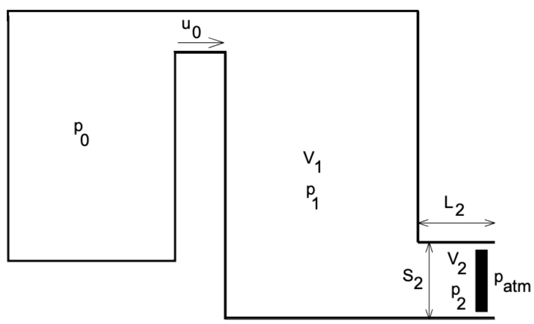
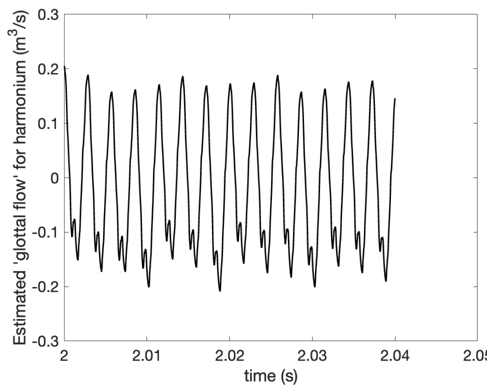


Figure 2: *Configuration of the Physical model.*

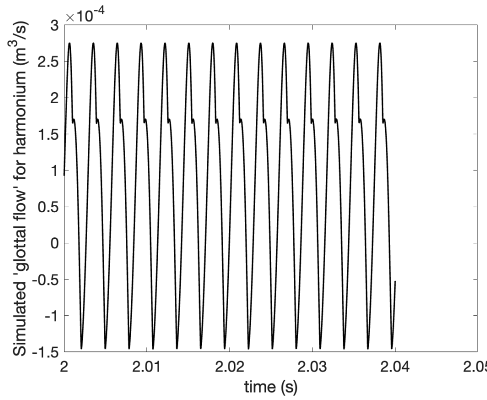
Figure 2 shows a schematic representation of the configuration of the physical model used to simulate the harmonium free-reed. The model assumes the region upstream of the reed (i.e. the reed chamber) to be a large volume V_1 where the pressure p_1 is uniform. The volume V_2, with cross-sectional area S_2 and length L_2, models the region near the reed where the air flows across the reed through a narrow jet. The reed itself was modeled as a sinusoidally oscillating lumped mass-spring-damper whose behavior is predominantly governed by the eigenfrequency (ω_0) and the quality factor (Q) parameters. The region downstream from the reed is exposed to atmospheric pressure. In the original Millot-Baumann model, the system was excited by the volume flow u_0 entering the volume V_1. In our adaptation, however, we added the chamber with pressure p_0 to represent the bellows pressure which indirectly controls the excitation signal u_0. With this change, we observed a better agreement between experimentally measured reed-chamber pressures and the corresponding p_1 values in the synthesis. It also allowed for the use of bellows pressure as a parameter to control the sound produced, like in a real harmonium. The governing equations for the system and the details of the numerical solution scheme are presented in [12, 4]. It should be noted that the bellows pressure p_0 and the reed eigenfrequency ω_0 are the only control parameters that vary while playing the harmonium. The note onset time and the stability of the numerical scheme was observed to be dependent on the other non-playing parameters.

3.2. Estimation of the wooden enclosure filter

The Iterative Adaptive Inverse Filtering (IAIF) algorithm developed by P. Alku [15] and implemented in the COVAREP Toolbox [16] can used to estimate the ‘glottal flow’ and the ‘vocal tract’ filter for a speech sound. For a harmonium sound, we expected the algorithm to estimate the analogous ‘reed airflow’ and ‘wooden enclosure’ filter respectively.



(a) *Estimated reed airflow.*



(b) *Simulated reed airflow.*

Figure 3: *A comparison of estimated vs simulated reed airflow.*

A comparison of the reed airflow estimated by the IAIF algorithm for a recorded harmonium note with the reed airflow simulated by the physical model is shown in Figs. 3a and 3b, respectively. The two signals display similar features. In particular, they show a discontinuity within each period that is assumed to occur when the reed passes from one side of its support plate to the other. The amplitude of the airflow signal estimated for the recorded sound is varying considerably, which is expected since the bellows pressure cannot be maintained perfectly constant in a real harmonium. Given the similarity, we proposed that the ‘vocal tract' filter estimated by the IAIF can be a good estimate of the 'wooden enclosure' filter.

4. SIGNAL MODEL AND IMPLEMENTATION IN PURE DATA

4.1. Source approximation

As mentioned in the previous section, the numerical stability for the physical model was dependent on the non-playing parameters, such as reed-chamber volume, reed clearance, etc. Hence, in order to change the playing frequency for a note, these parameters had to be changed in addition to the reed eigenfrequency (ω_0) and they had to be determined by manual trial-and-error, which is not possible in a live-performance context. Moreover, it was observed that the filter has a much stronger influence on the synthesized sound timbre than the source. Hence, a simplified signal model capable of approximately generating the reed source signal at the different playing frequencies would produce perceptually similar results while eliminating the problems of numerical stability and computation cost involved in the physical model.

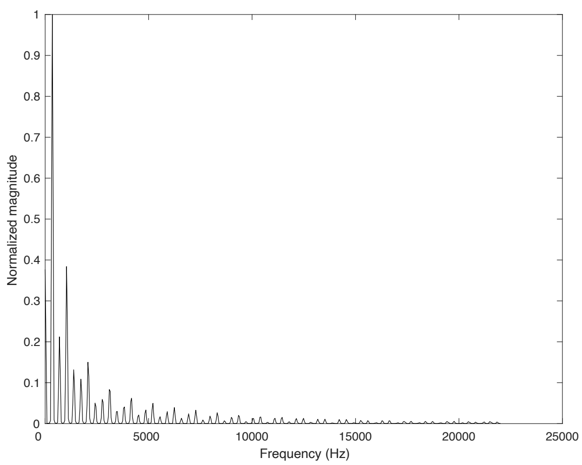


Figure 4: *Normalized LTAS for the reed source signal simulated using physics model.*

The normalized Long Term Average Spectrum (LTAS) for an 'F4' note (349 Hz) simulated by the physical model can be seen in Fig. 4. A band-limited signal with harmonic weights equal to the peak heights in the normalized LTAS was used to model the reed source signal. In our Pure Data implementation (Fig. 6), this was achieved by using a 'sinesum' message with the harmonic weights of the first 37 peaks in the normalized LTAS to populate a wavetable. Notes with different frequencies were synthesized by reading the wavetable at the particular frequencies.

4.2. Filter approximation

The all-pole filter estimated by the IAIF algorithm had 49 coefficients. However, the magnitude filter response for different harmoniums tested was observed to have 8-9 peaks. Hence, a close approximation of the IAIF estimated filter using a series of 10 second order sections (i.e. 10 biquad elements in a cascade in the Pure Data implementation) was deemed feasible. In MATLAB, the 'stmcb' algorithm [17] was used to estimate an approximate filter with 10 poles and 10 zeros and the 'tf2sos' function was then used to determine the second order filter coefficients. Also, a smooth frequency response curve passing through all the peaks and valleys was obtained using the 'pchip' algorithm [18] and the 'yulewalk' algorithm [19] was used to construct a filter with that frequency response. Although there was a larger error in the second approach for the filter approximation, the error was observed to be perceptually insignificant in the synthesis while allowing for the parametrization of the filter on the basis of just 9 peak and 9 valley points. Such a parametrization would be helpful to construct arbitrary frequency responses that can be used to synthesize harmonium sounds with a variety of timbres. The IAIF filter response and its approximations using the two methods described can be seen in Fig. 5.

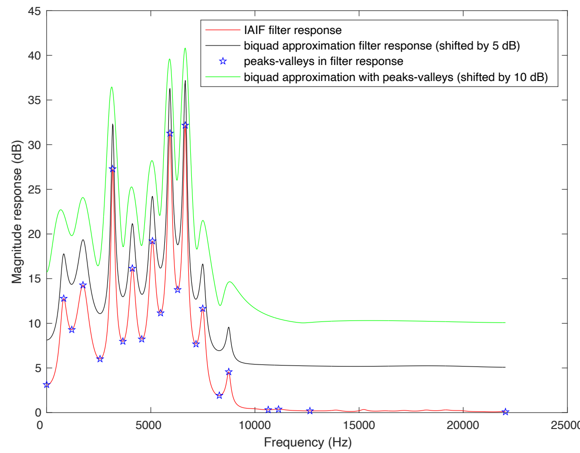


Figure 5: *Frequency response of estimated and approximated wooden enclosure filters for a concert quality harmonium.*

4.3. Effect of bellows pressure

The primary role of the bellows pressure is to modulate the amplitude of the sound. Additionally, in the physical model as well as in the case of recorded harmonium sounds, it is observed that the note fundamental frequency (f0) slightly varies when the bellows pressure is changed. Although the variation is very small (typically less than 1-2 Hz), it is still perceptible and is used by harmonium players to create a perceptual effect of continuous pitch glides through specific bellowing techniques. The most prominent effect of this behaviour is the slight increase in playing frequency as the air leaks out of the harmonium, thus reducing the bellows pressure. In the Pure Data implementation, the bellows pressure is used to control the amplitude envelope of the sound. In addition, it is added to the 'f0' frequency to change the rate at which the wavetable is read to mimic the frequency variation described.

5. CONCLUSIONS

The paper describes a signal model for synthesizing Indian hand harmonium sounds. The signal model uses a source-filter structure similar to a previously developed physical model. The source

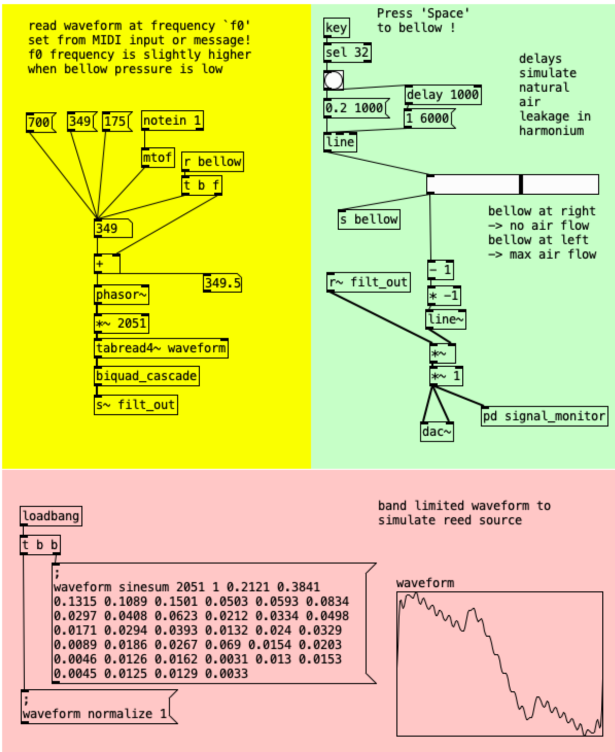


Figure 6: *Pure Data patch to implement the signal model.*

is constructed as a band-limited waveform that has similar spectral characteristics as the physical model. Simplifications are suggested to parametrize the filter on the basis of the prominent peaks in the filter frequency response. Different timbres of the harmonium can be realized by hand-crafting or estimating the filter parameters. The simplified model implemented in Pure Data presents the possibility of real-time synthesis and live performance using the virtual instrument.

6. ACKNOWLEDGMENTS

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