Handheld Acoustic Filter Bank for Musical Control

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ABSTRACT

This paper introduces the design of a handheld musical input device that produces control data by measuring, and analyzing, the resonances of carefully tuned pipes. The device provides input control information to several virtual reed instruments running in parallel, responsible for producing the sound. Inspired by the khaen, a musical instrument from Northeast Thailand and Laos, the controller consists of a row of acoustic tubes, with finger holes that change the tube's resonance when covered. Each tube is equipped with both a microphone recording the change in pressure variations at a set location along the tube. The mic outputs are mixed, and input to the computer via the mic level audio ports, allowing the controller to interface very reliably (and conveniently) to most laptop computers.

Keywords

khaen, sound synthesis control, mapping, musical acoustics

1. INTRODUCTION

Western "breath-driven" musical instruments [1] typically make use of a mouthpiece and a single bore that can take on a variety of shapes from cylindrical tubes to conical or flared horns, or some piecewise combination thereof. If a player is to change the pitch on such an instrument, some mechanism, such as toneholes or nested sliding tubes, must be in place for changing the bore's effective acoustic length[2]. These mechanisms have served musicians well over the course of western music history, so well in fact, that their ubiquitousness continues to pervade our paradigms for new musical input devices. Though these devices have offered substantial possibilities for controlling tone [4, 10, 11, 3, 6], because they are monophonic they are far less suitable as controllers for more general musical needs.

Following research on the generalized virtual reed model [12], there existed a need for an input device that would allow for the control of an arrangement, in series or parallel, of multiple pressure-controlled valves. As is the tradition in the development of new music controllers, the author began by looking at existing tried and true control mechanisms that would meet the requirements. The *khaen*, a breath-driven instrument from Northeast Thailand and Laos, consisting of two rows tuned acoustic pipes which may be played independently or simultaneously, served as the model.

Many new musical controllers are developed independently from the computer sound synthesis algorithms they control. This disconnect has made it increasingly difficult to develop mapping strategies for parameter laden physical modelling synthesis algorithms, for which the input control parameters and the synthesis parameters are very strongly linked. This research therefore also attempts to address the notion of *re-linking* acoustics to controller by measuring and analyzing the resonances of carefully tuned pipes. That is, rather than using sensors to measure the user's input directly, a microphone measures the state of the pipe as it is altered by the user.

The benefits of this approach are twofold: 1) the user isn't forced to interact directly with bulky, often overly wired and flaky electronics that tend to break, and frequently need replacing from over-use and 2) the controller can directly interface with the audio ports on a laptop computer, without requiring a micro-controller and a lot of extra circuitry for data acquisition—returning the concept of portability to musicians who use laptop computers during performance.

The idea for this controller was inspired both by ongoing research on the Khaen, as well as Bernard Chouet's approach of detecting "long period events" for predicting imminent volcano eruptions [8]. Both make use of resonance phenomena.

2. KHAEN ACOUSTIC DESIGN

The *khaen* is a free reed, mouth-organ style instrument found in the Northeast region of Thailand (Issan) and Laos. Very similar to the Japanese *sho*, or the Chinese *sheng*, it consists of two rows of bamboo tubes decreasing in length, each tube having its own reed, and being responsible for a single pitch lying within a diatonic scale [7]. The player supports the instrument upright in front of the mouth, with the

[1]The term "breath-driven" is used to distinguish from *wind* or *wind-driven* instruments, which would include organs and accordions, among others.
[2]Some reed driven instruments, such as the digiridoo, rely exclusively on the mechanical resonance of the reed to alter the sounding pitch, though their pitch range is rather limited as a result.

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Figure 1: Mr. Khene, a khaen maker from Roi Et, Northeast Thailand, displays the tuning vents in the pipes.

windchest resting comfortably between the palms of prayer positioned hands, with holes within the fingers reach. No sound is produced from a particular pipe unless its finger hole is covered. Blowing, or inhaling, through a mouth hole provides a pressure or vacuum inside the air chamber surrounding the reeds [2], creating a differential pressure across the reed that forces it into oscillation.

Pipes are inserted into a windchest, called the *tao*, so they form two rows of anywhere from 6-10—but usually 8—pipes (see Figure 1). The perfect skyline arrangement of the pipes is purely aesthetic as the actual pipe lengths are determined using vent holes placed a distance L apart, approximately 3/4 L above and approximately 1/4 L below the position of the reed (see Figure 1). The distance L is therefore, the effective acoustic length of the pipe when the finger hole is covered. An open finger hole, melted into the bamboo using a heated iron awl, drastically reduces the acoustic length of the pipe, altering its resonance so that it no longer couples strongly with the reed.

Finger holes in the khaen do not function as toneholes in single bore instruments. Though the small hole alters the effective acoustic length of the pipe if left uncovered, it also destroys the pipe resonance and prevents the reed from vibrating. Therefore it is possible to blow into the instrument and not produce any sound. The player covers only the holes for the pipe(s) that should sound, and may play several notes simultaneously. The rubbery material, called *kisoot*, that seals the pipes and the *tao*, may also be used to close finger holes, creating a fuller sound with drones or pedal notes.

The arrangement of the pipes and the proximity of their finger holes make it relatively easy to play notes in rapid succession (with a little practice). Since very little finger pressure is needed to play a pipe, it is fairly easy to flutter the fingers over the holes and, again with practice, to play with speed and virtuosity. This is an aspect of the khaen that would be very difficult to capture using force sensing resistors but that has been retained in this design. Khaen players will also say that the texture of the hole outline beneath the finger provides an important cue for locating its position (which, is important given the holes are not visible during performance) [1]. This feature is also retained in this design. An extension of the Khaen within the capability of the controller, is the implementation of partial hole coverage, rather than the binary choice of open or closed. This may be less helpful to traditional players, but certainly offers more flexibility to those who wish to extend the instrument [9].

3. ACOUSTIC FILTER BANK FOR INPUT PARAMETERS

The controller is designed to serve as an input device for playing a parallel arrangement of generalized reed synthesis models [12]. Clearly the arrangement of pipes, one for each sounding pitch, is a particularly useful paradigm for controlling several virtual reeds. The controller however, also incorporates the Khaen's use of resonance as a mechanism for control.

Like the khaen, the controller consists of a bank of carefully tuned pipes with finger holes placed within the reach of the player's hands, and a mouthpiece supplying pressure variations—though this research focuses on capturing the fingering of the player rather than his/her blowing technique (which in reality has very little variation since the player must blow rather hard to produce sound).

Each tube n is cut to a length L_n, and closed at either end to ensure a "perfect" reflection—a speaker seals one end while a piece of acrylic seals the other. A small finger hole is placed in the wall of the tube at the end opposite the speaker, effectively opening that end (see Figure 2). Closing the finger hole changes the configuration of the pipe from one that is closed-open to one that is closed-closed.

Notice from Figure 2 that though displacement of air is greatest at an *open end*, the pressure variation is maximum at a *closed end*. For the first harmonic of a closed-closed tube, there is maximum pressure variation at both ends, with a pressure null at the center. For the second harmonic of the same configuration, adding a node and antinode produces a standing wave pattern with a pressure maximum in the center of the tube. Driving the tube at a frequency equal to this second harmonic, and placing a microphone at the center of the tube, will measure maximum pressure variation when the hole is covered. Uncovering the hole reverts the configuration back to a closed-open tube with an altogether different (lower) resonance, with the second harmonic (and every even harmonic) missing (see Table 1, [5] or any acoustics text for calculating tube resonant frequencies). As a result, there is a considerable increase in the sound pressure level recorded by the microphone if the hole is covered.

A signal is generated from the computer that combines the appropriate driving frequencies (see frequencies from Table 2). This signal is output via the speakers at the end of each tube, and is effectively filtered by a bank of acoustic filters, that is, each pipe acts as a filter, boosting or attenuating only those frequencies corresponding to one of its resonant modes. The output from each of the microphones is added together using a very simple mixing circuit (see Appendix), and the sum is input back into the computer's mic level audio inputs, scanned and analyzed to determine which pipes have been selected for sound production.

The sound produced by the tubes must be within the audio range (not exceeding the Nyquist limit) so that the input and output can be handled using the computer's audio

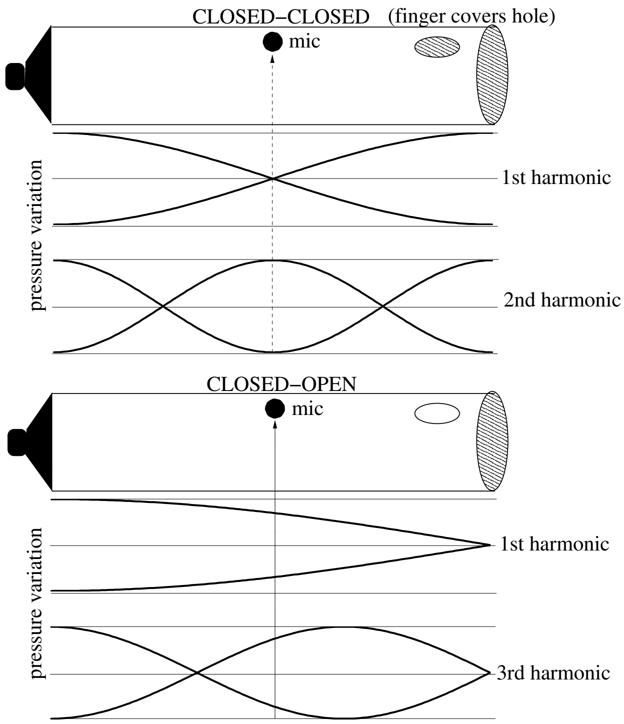

Figure 2: The first two modes of vibration, showing pressure variation maxima and minima, and standing wave patterns, for closed-closed acoustic tubes (resulting from the the finger covering the hole), and closed-open tubes (where the finger hole is open).

Boundary	Harmonic k	Wavelength λ	Frequency $f_k = c/\lambda$
Closed-Closed	1	$2L$	$f_1 = 425$ Hz
	2	L	$f_2 = 850$ Hz $= 2f_1$
Closed-Open	1	$4(L + 0.61a)$	$f_1 \approx 206$ Hz
	3	$\frac{4}{3}(L + 0.61a)$	$f_3 \approx 618$ Hz $= 3f_1$

Table 1: Modes of vibration for a closed-closed and closed-open pipe of length $L = 40$ cm and radius $a = 2$ cm, computed using speed of sound, $c = 340$. In an actual pipe, pressure variations drop to zero slightly beyond an open end, effectively increasing the acoustic length for this condition by approximately $0.61a$, where a is the radius of the pipe.

		Closed-Closed		Closed-Open	
pipe	L (cm)	f_1 (Hz)	f_2 (Hz)	f_1 (Hz)	f_3 (Hz)
1	20	1700.0	**3400.0**	0757.6	2272.7
2	12	1416.7	**2833.3**	0643.0	1928.9
3	14	1214.3	**2428.6**	0558.5	1675.4
4	18	0944.4	**1888.9**	0442.2	1326.7
5	20	0850.0	**1700.0**	0400.6	1201.7
6	22	0772.7	**1545.5**	0366.1	1098.2
7	30	0566.7	**1133.3**	0272.3	0816.8
8	35	0485.7	**0971.4**	0234.7	0704.0

Table 2: Tube lengths L (in centimeters), closed-closed (Cl) or closed-open (Op) and frequencies f_0 and f_1 (in Hz) for 8 pipes.

ports. The closed-closed tube configuration was chosen to limit sound radiation to the player's ear when a tube resonates, as the user holds the instrument (at the mouth) in fairly close proximity to the ears. Since all the tubes receiving the same excitation signal, the tubes had to be given a variety of lengths, shown in Table 2, to ensure that the 2nd harmonic of one closed-closed tube doesn't excite a harmonic of another closed-open tube.

Selecting a tube on the controller activates a synthesis model of a generalized reed, fully detailed in [12], producing the instrument sound. Though the controller currently doesn't allow the user to modify many of the model's continuously variable input parameters, it solves the problem of playing several reeds simultaneously. For some reed types, particularly free reeds, where only a limited number of combinations of input parameters produced satisfying sound quality, the ability to control several models was all that was required.

4. CONCLUSIONS

With the surfacing of new controllers and computer sound synthesis algorithms, there has been a clear disconnect between the device that produces the sound and the device with which the player/musician interacts. Various sensors may be used to capture human input that has very little, if anything, to do with the mechanism or acoustics of the produced sound. Though this would seemingly increase the possibilities for the design and development of new controllers unfettered by the demands of the acoustic systems they control, in actual practice it seems to have been more of a hindrance in the development of quality musical instruments with intuitive mapping.

This work presents the design of an instrument that uses the acoustic information of the controller, and remaps it for parameter control of a computer synthesis model—effectively preserving, but without being limited or constrained by, the acoustics on which it is based.

5. ACKNOWLEDGMENTS

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Figure 3: Mr. Samong plays a Khaen using "extended" techniques.

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APPENDIX

A. MIXER DESIGN

The microphone signals must be added together before connecting them to the mic level input of the laptop. Simpling placing the speaker outputs in parallel will overload the outputs and likely increase the distortion in the sound. A better approach is to create a very simple mixer circuit, which simply involves inserting the appropriate resistor (see Figure 4) at the output before summing them together.

Since the computer's mic level audio input is stereo, two mixers can be used to sum the microphone signals from each row, and then input into the computer via the left and right channel.

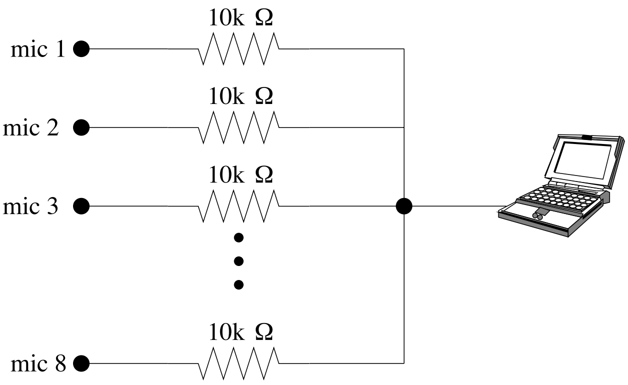

Figure 4: A simple mixing circuit for up to 8 input channels.