Model-based Synthesis of the Ud and the Renaissance Lute

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Abstract

We describe techniques for sound synthesis of the ud and the Renaissance lute using the physical modeling approach. In the present model, novel methods have been used for the design of the loop filters, as well as for the implementation of the glissando effect. With its realistic synthetic sounds, our system can provide new tools and a test-bed platform for researchers, educators and composers of both the Western world and of the Eastern traditions.

1 Introduction

Model-based synthesis is one of the most recent paradigms in sound synthesis (Smith 1992; Smith 1996). The basic principle of the method, i.e., simulating the sound production mechanisms, can be applied to synthesize a broad range of traditional musical instruments, as well as to create entirely virtual ones (Cook 1997). The availability of the open-source tools, such as STK, allows the musicians and researchers to implement and extend the basic algorithms (Cook and Scavone 1999).

The model-based sound synthesis of plucked string instruments has been an ongoing interest of the field (Jaffe and Smith 1983; Smith 1996; Karjalainen, Välimäki, and Tolonen 1998). The design, implementation, and control of a recent real-time system has been reported in (Laurson et al. 2001). This paper documents our achievements in simulating different plucked string instruments, namely the Turkish ud (mentioned as the ud in the following) and the Renaissance lute.

The structure of this paper is as follows. In Sec. 2, we briefly summarize the properties of the ud and the lute. In Sec. 3, we introduce the common synthesis model and discuss the model calibration, the extraction of the most prominent body modes, the design of second-order loop filters, and implementation of the glissandi. In Sec. 4, we summarize the control of the lute models using enriched notation software that is customized for the control of the physical models. Finally, Sec. 5 is devoted to a discussion about our motivations and goals, and potential application areas of the described techniques, as well as the general use of the overall synthesis system.

2 Properties of the Ud and the Lute

Both the ud and the Renaissance lute are members of the short-necked lute family. Historically, both the lute and the Turkish ud are descendants of the Arabic instrument *al'ud* that has a wooden soundboard and a characteristic staved, wood-vaulted back design. The lute underwent continuous adaptation and evolution according to the requirements of European music. European lutes are now divided into three general categories; the medieval lute, the Renaissance lute, and the Baroque lute. The lute part of the present study focuses on the Renaissance tenor lute.

2.1 Properties of the Ud

The Turkish ud has five courses tuned in unison, and a single bass string used as chanterelle. The tuning used in this study is $D_1, A_1, B_1, E_2, A_2, D_3$. The ud is a fretless instrument in order to reproduce the micro tones required by the Turkish modal system *makam* (Signell 1977). The two treble courses are made of nylon and the others of metal wound with fine silk or nylon thread. The ud is played with a plectrum (*mizrap*). The soundboard has three latticed sound holes; one large and two small ones.

2.2 Properties of the Renaissance Lute

The Renaissance lute has seven string groups, and it is played with fingers. All the strings except the first one are in pairs that are tuned in unison. Different sizes of the Renaissance lute form a consort (soprano, alto, tenor, and bass). The most common instrument for solo playing is the tenor lute. This study focuses on the tenor lute with the following tuning: $D_1, G_1, C_2, F_2, A_2, D_3, G_3$. There are 8 frets on the neck, and several frets may be placed on the body, especially

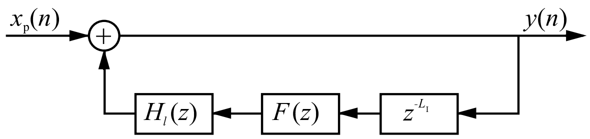

Figure 1: The basic string model.

for treble courses. The soundboard has a single latticed sound hole.

3 Synthesis Model

The string models of the ud and the lute are based on digital waveguide approach (Smith 1992). A single-polarization string model (Karjalainen, Välimäki, and Tolonen 1998) is used for each string. The basic string model $S(z)$ is shown in Fig. 1. It consists of a delay line z^{-L_1}, a loop filter $H(z)$, and a fractional delay filter $F(z)$. The loop filter simulates the frequency-dependent losses of a real string, and the fractional delay filter is used for fine-tuning of the string model. The input $x(n)$ is an excitation signal that contains the body modes (except the most prominent ones, as discussed further below) in accordance with the commuted waveguide synthesis approach (Smith 1993; Karjalainen, Välimäki, and Jánosy 1993).

Fig. 2 depicts the common instrument model used in this study. It contains 11 string models (a single $S(z)$ is shown in the figure), together with a coupling matrix that controls the sympathetic vibration of the strings. The model parameters and excitation signals are obtained by analyzing the recordings of the instruments as described in (Laurson et al. 2001), and stored separately for each instrument. The excitation signals are further processed as described below in Sec. 3.1, and the most prominent body modes are synthesized in parallel, using separate body resonators. The special effects (plectrum and finger scratches, body tappings, etc.) are stored in a separate database and their gains are individually controlled.

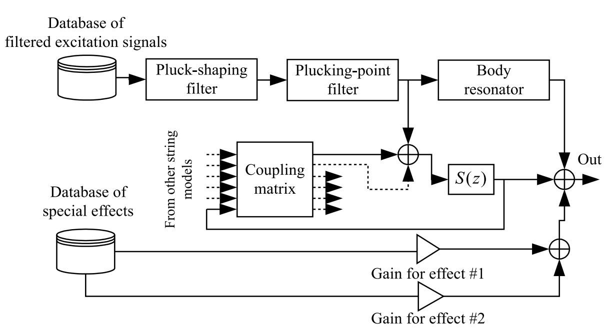

Figure 2: The lute instrument model.

3.1 Resynthesis of the Body Modes

We conducted a series of acoustic measurements on the ud and lute bodies, and identified the most prominent modes ($f_1 = 113$ Hz for the ud, and $f_1 = 144Hz$ for the lute). These prominent modes are removed from the excitation signals as described in (Karjalainen et al. 2000), i.e, using the following notch filter

$$H_n(z) = \frac{A(z)}{A(z/c)} = \frac{1 + a_1 z^{-1} + a_2 z^{-2}}{1 + c a_1 z^{-1} + c^2 a_2 z^{-2}} \qquad (1)$$

where $c = 1 - \epsilon$ is a spectrum-smoothing parameter, and ϵ is a very small positive number. The a parameters of the notch filter are obtained by analyzing the frequency and the bandwidth of the body resonance in the excitation signals. At the synthesis step, the following second-order resonator is used to reconstruct the body modes:

$$H_r(z) = g \frac{b_0 + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-1}} \qquad (2)$$

where the denominator polynomial equals to the numerator of the notch filter in Eq. (1), g is a scaling parameter that controls the magnitude of the body resonance, and $b_2 = b_0$. Note that this form of the second-order resonator has sharp notches in the DC and Nyquist frequencies, and a single tunable resonance peak. The following example demonstrates the method.

An extracted lute excitation signal (first string, open position, f_0 = 389.9 Hz) is analyzed and the notch filter coefficients are obtained. The original spectrum of the signal is shown at the top-left part of Fig. 3. The body resonance is filtered by the notch filter, and the spectrum of the filtered signal is shown at the top-right part of the figure. The plot at the bottom compares the spectrum of the reconstructed resonance with the original spectrum. In many cases, the reconstructed resonance had a pronounced perceptual similarity with the original one. Besides reducing the memory costs by shortening the stored excitation signals, the described approach allowed us to experiment with cross-synthesis, i.e., using the ud excitation signal for the lute, and vice versa.

3.2 Higher-Order Loop Filters

The loop filter used in this study is the second-order IIR filter (*biquad*) with the following transfer function:

$$H_l(z) = g \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}} \qquad (3)$$

where g is a gain parameter that controls the DC gain of the filter. Since the loop filter is placed inside of a feedback loop,

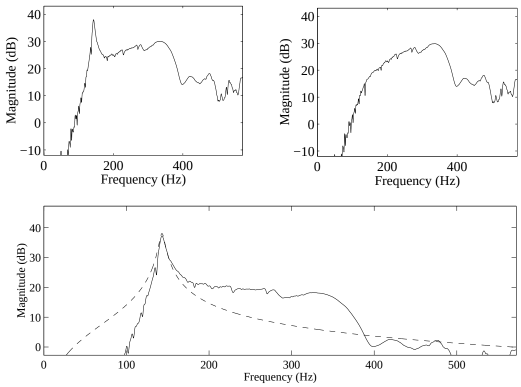

Figure 3: The extraction of a body mode from a lute excitation signal. The body mode occurs at $f_1 = 144$ Hz (top-left), and it is effectively removed by the notch filter of Eq. (1) (top-right). The resonator of Eq. (2) reconstructs the body resonance (bottom).

there is a stability constraint on the filter magnitude response, i.e., $|H_l(e^{\jmath\omega})| < 1$ on the entire frequency range.

The problem of arbitrary-order recursive filter that approximates a given complex transfer function has been extensively studied in (Smith 1983). However, there is still no method that guarantees both the accuracy of the approximation and the stability. Moreover, we experimentally observed that an analysis routine exhibits large deviations in the extracted decay characteristics of the plucked string tones (Erkut 2001). This practically means that the design techniques have to be extended to regularize the deviations in the desired transfer function.

A novel technique for the loop filter design has been described in (Erkut 2001). As an initial step, this technique approximates the extracted decay rate function $\sigma(\omega)$ (inverse of the decay times τ_k of the harmonics) by low-order algebraic polynomials. The order m of the approximation is determined using analytical model-order selection criteria, i.e, by penalizing the high-order polynomials. The truncated polynomials are converted to the target magnitude response $|H_m(\omega)|$ using the following relation

$$\begin{aligned} \sigma(\omega) &\approx c_1 + c_3\omega^2 + \ldots + c_{2m+1}\omega^{2m} \\ &= -f_0 \log(|H_{\mathrm{m}}(\omega)|) \end{aligned} \tag{4}$$

Finally, the loop filter is designed by the modified Yule-Walker method (Friedlander and Porat 1984) using the regularized magnitude response $|H_m(\omega)|$. Fig. 4 shows the measured gains of an analyzed lute tone and the designed biquad filter magnitude response. In many cases, the match is satisfactory, and the stability is achieved.

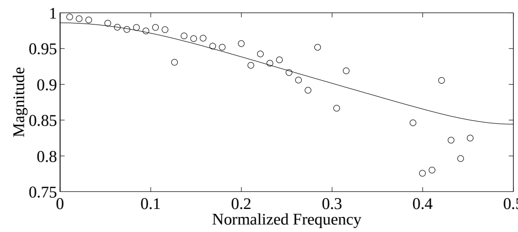

Figure 4: The biquad filter magnitude response (solid curve) compared with the desired gains of the harmonics (circles) of a lute tone (first string, third fret, $f_0 = 464$ Hz).

3.3 Glissandi

As mentioned in Sec. 2, the lute is a fretted instrument, whereas the ud is a fretless one. This structural difference manifests itself in the measured glissandi regimes. Fig. 5 shows the fundamental frequency (top) and the instantaneous energy (bottom) as a function of time during a typical lute glissando. The frequency trajectory has a step-like character; each time a different fret is reached, there is an increase in the signal energy. The reason for the increase is that the string is re-excited as a result of the interaction between the finger, the string and the fret. In our implementation, we simulated this behavior by using adequately scaled excitation signals.

Fig. 6 shows the frequency (top) and the energy (bottom) during a typical ud glissando. In this case, the re-plucking is absent since the instrument is fretless. The increased decay of energy is due to lossy termination of the string by the flesh of the finger, and it is simulated by lowering the loop filter gain during the synthetic glissando. Since the glissandi are the backbones in ud playing technique, the fundamental frequency trajectories and decay characteristics of the recorded samples provide the templates for a realistic simulation of the glissandi.

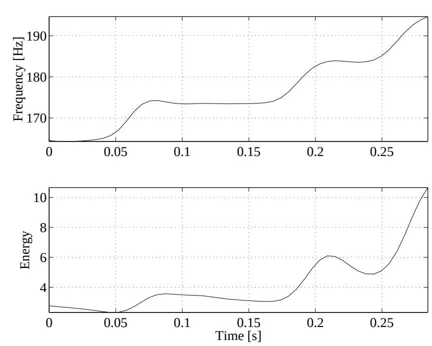

Figure 5: A typical example of a lute glissando.

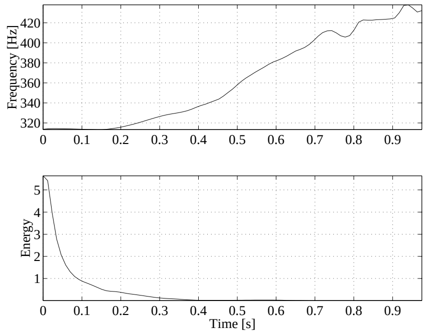

Figure 6: A typical example of an ud glissando.

4 Control

The system is controlled with a self-made sequencer, which uses an enriched notation format and contains specific additions customized for realistic and expressive model-based synthesis of string instruments. The mode-based tuning of the ud is obtained by special add-ons, and some basic makams have been implemented. These add-ons include a set of special accidental symbols that are used to denote the micro tones in the respective makam scale. Our notation system allows to change the timing of the notes by special rubato tempo functions in order to capture the improvisatory and expressive nature of a performance. Also the notation adds special symbols that allow the performance to include some of the idiomatic playing styles used by an ud player such as glides and hits with the plectrum on the body of the instrument.

5 Conclusions and Future Plans

Our motivations in this research were multiple-fold. The lutes individually contributed into our existing repertoire of string instruments. Especially, the ud synthesis model and its control provided new challenges for our temperament-based system. Moreover, the second-order loop filters and body resonators were the recent extensions in the implementation sense. We also experimented with some pairings between the lute models and the control structure. These experiments included playing the typical repertoire of one instrument using the synthesis model of the other.

Besides its musical uses, we believe that such a system is especially beneficial for the research of Middle Eastern music. Given that there is no solid theoretical foundation of the modal structure, a system that allows the researcher to experiment with different theories may accelerate the research in this field. During the ICMC conference, we will present musical examples performed using the system. Both Middle Eastern and Renaissance compositions will be included in the demonstration. These sound examples will also be available at the following URL:

`http://www.acoustics.hut.fi/publications/`.

6 Acknowledgments

This work has been financially supported by the Academy of Finland and Foundation of Jenny and Antti Wihuri.

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