

Sound Synthesis of the Harpsichord Using a Computationally Efficient Physical Model

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A sound synthesis algorithm for the harpsichord has been developed by applying the principles of digital waveguide modeling. A modification to the loss filter of the string model is introduced that allows more flexible control of decay rates of partials than is possible with a one-pole digital filter, which is a usual choice for the loss filter. A version of the commuted waveguide synthesis approach is used, where each tone is generated with a parallel combination of the string model and a second-order resonator that are excited with a common excitation signal. The second-order resonator, previously proposed for this purpose, approximately simulates the beating effect appearing in many harpsichord tones. The characteristic key-release thump terminating harpsichord tones is reproduced by triggering a sample that has been extracted from a recording. A digital filter model for the soundboard has been designed based on recorded bridge impulse responses of the harpsichord. The output of the string models is injected in the soundboard filter that imitates the reverberant nature of the soundbox and, particularly, the ringing of the short parts of the strings behind the bridge.

Keywords and phrases: acoustic signal processing, digital filter design, electronic music, musical acoustics.

1. INTRODUCTION

Sound synthesis is particularly interesting for acoustic keyboard instruments, since they are usually expensive and large and may require amplification during performances. Electronic versions of these instruments benefit from the fact that keyboard controllers using MIDI are commonly available and fit for use. Digital pianos imitating the timbre and

features of grand pianos are among the most popular electronic instruments. Our current work focuses on the imitation of the harpsichord, which is expensive, relatively rare, but is still commonly used in music from the Renaissance and the baroque era. [Figure 1](#) shows the instrument used in this study. It is a two-manual harpsichord that contains three individual sets of strings, two bridges, and has a large soundboard.



FIGURE 1: The harpsichord used in the measurements has two manuals, three string sets, and two bridges. The picture was taken during the tuning of the instrument in the anechoic chamber.

Instead of wavetable and sampling techniques that are popular in digital instruments, we apply modeling techniques to design an electronic instrument that sounds nearly identical to its acoustic counterpart and faithfully responds to the player's actions, just as an acoustic instrument. We use the modeling principle called commuted waveguide synthesis [1, 2, 3], but have modified it, because we use a digital filter to model the soundboard response. Commuted synthesis uses the basic property of linear systems, that in a cascade of transfer functions their ordering can be changed without affecting the overall transfer function. This way, the complications in the modeling of the soundboard resonances extracted from a recorded tone can be hidden in the input sequence. In the original form of commuted synthesis, the input signal contains the contribution of the excitation mechanism—the quill plucking the string—and that of the soundboard with all its vibrating modes [4]. In the current implementation, the input samples of the string models are short (less than half a second) and contain only the initial part of the soundboard response; the tail of the soundboard response is reproduced with a reverberation algorithm.

Digital waveguide modeling [5] appears to be an excellent tool for the synthesis of harpsichord tones. A strong argument supporting this view is that tones generated using the basic Karplus-Strong algorithm [6] are reminiscent of the harpsichord for many listeners.¹ This synthesis technique has been shown to be a simplified version of a waveguide string model [5, 7]. However, this does not imply that realistic harpsichord synthesis is easy. A detailed imitation of the properties of a fine instrument is challenging, even though the starting point is very promising. Careful modifications to the algorithm and proper signal analysis and calibration routines are needed for a natural-sounding synthesis.

The new contributions to stringed-instrument models include a sparse high-order loop filter and a soundboard

model that consists of the cascade of a shaping filter and a common reverb algorithm. The sparse loop filter consists of a conventional one-pole filter and a feedforward comb filter inserted in the feedback loop of a basic string model. Methods to calibrate these parts of the synthesis algorithm are proposed.

This paper is organized as follows. Section 2 gives a short overview on the construction and acoustics of the harpsichord. In Section 3, signal-processing techniques for synthesizing harpsichord tones are suggested. In particular, the new loop filter is introduced and analyzed. Section 4 concentrates on calibration methods to adjust the parameters according to recordings. The implementation of the synthesizer using a block-based graphical programming language is described in Section 5, where we also discuss the computational complexity and potential applications of the implemented system. Section 6 contains conclusions, and suggests ideas for further research.

2. HARPSICHORD ACOUSTICS

The harpsichord is a stringed keyboard instrument with a long history dating back to at least the year 1440 [8]. It is the predecessor of the pianoforte and the modern piano. It belongs to the group of plucked string instruments due to its excitation mechanism. In this section, we describe briefly the construction and the operating principles of the harpsichord and give details of the instrument used in this study. For a more in-depth discussion and description of the harpsichord, see, for example, [9, 10, 11, 12], and for a description of different types of harpsichord, the reader is referred to [10].

2.1. Construction of the instrument

The form of the instrument can be roughly described as triangular, and the oblique side is typically curved. A harpsichord has one or two manuals that control two to four sets of strings, also called registers or string choirs. Two of the string choirs are typically tuned in *unison*. These are called the 8' (8 foot) registers. Often the third string choir is tuned an octave higher, and it is called the 4' register. The manuals can be set to control different registers, usually with a limited number of combinations. This permits the player to use different registers with left- and right-hand manuals, and therefore vary the timbre and loudness of the instrument. The 8' registers differ from each other in the plucking point of the strings. Hence, the 8' registers are called 8' back and front registers, where “back” refers to the plucking point away from the nut (and the player).

The keyboard of the harpsichord typically spans four or five octaves, which became a common standard in the early 18th century. One end of the strings is attached to the nut and the other to a long, curved bridge. The portion of the string behind the bridge is attached to a hitch pin, which is on top of the soundboard. This portion of the string also tends to vibrate for a long while after a key press, and it gives the instrument a reverberant feel. The nut is set on a very rigid wrest plank. The bridge is attached to the soundboard.

¹The Karplus-Strong algorithm manages to sound something like the harpsichord in some registers only when a high sampling rate is used, such as 44.1 kHz or 22.05 kHz. At low sample rates, it sounds somewhat similar to violin pizzicato tones.

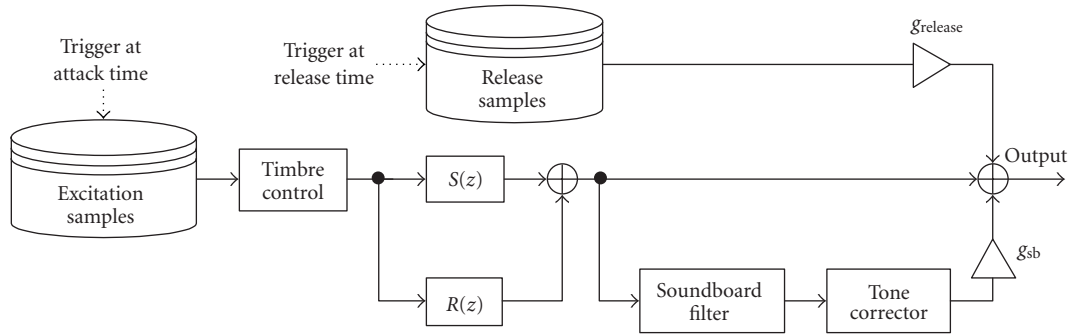


FIGURE 2: Overall structure of the harpsichord model for a single string. The model structure is identical for all strings in the three sets, but the parameter values and sample data are different.

Therefore, the bridge is mainly responsible for transmitting string vibrations to the soundboard. The soundboard is very thin—about 2 to 4 mm—and it is supported by several ribs installed in patterns that leave trapezoidal areas of the soundboard vibrating freely. The main function of the soundboard is to amplify the weak sound of the vibrating strings, but it also filters the sound. The soundboard forms the top of a closed box, which typically has a rose opening. It causes a Helmholtz resonance, the frequency of which is usually below 100 Hz [12]. In many harpsichords, the soundbox also opens to the manual compartment.

2.2. Operating principle

A plectrum—also called a quill—that is anchored onto a jack, plucks the strings. The jack rests on a string, but there is a small piece of felt (called the damper) between them. One end of the wooden keyboard lever is located a small distance below the jack. As the player pushes down a key on the keyboard, the lever moves up. This action lifts the jack up and causes the quill to pluck the string. When the key is released, the jack falls back and the damper comes in contact with the string with the objective to dampen its vibrations. A spring mechanism in the jack guides the plectrum so that the string is not replucked when the key is released.

2.3. The harpsichord used in this study

The harpsichord used in this study (see Figure 1) was built in 2000 by Jonte Knif (one of the authors of this paper) and Arno Pelto. It has the characteristics of harpsichords built in Italy and Southern Germany. This harpsichord has two manuals and three sets of string choirs, namely an 8' back, an 8' front, and a 4' register. The instrument was tuned to the Vallotti tuning [13] with the fundamental frequency of A_4 of 415 Hz.² There are 56 keys from G_1 to D_6 , which correspond to fundamental frequencies 46 Hz and 1100 Hz, respectively, in the 8' register; the 4' register is an octave higher, so the corresponding lowest and highest fundamental frequencies are about 93 Hz and 2200 Hz. The instrument is 240 cm long

and 85 cm wide, and its strings are all made of brass. The plucking point changes from 12% to about 50% of the string length in the bass and in the treble range, respectively. This produces a round timbre (i.e., weak even harmonics) in the treble range. In addition, the dampers have been left out in the last octave of the 4' register to increase the reverberant feel during playing. The wood material used in the instrument has been heat treated to artificially accelerate the aging process of the wood.

3. SYNTHESIS ALGORITHM

This section discusses the signal processing methods used in the synthesis algorithm. The structure of the algorithm is illustrated in Figure 2. It consists of five digital filters, two sample databases, and their interconnections. The physical model of a vibrating string is contained in block $S(z)$. Its input is retrieved from the excitation signal database, and it can be modified during run-time with a timbre-control filter, which is a one-pole filter. In parallel with the string, a second-order resonator $R(z)$ is tuned to reproduce the beating of one of the partials, as proposed earlier by Bank et al. [14, 15]. While we could use more resonators, we have decided to target a maximally reduced implementation to minimize the computational cost and number of parameters. The sum of the string model and resonator output signals is fed through a soundboard filter, which is common for all strings. The tone corrector is an equalizer that shapes the spectrum of the soundboard filter output. By varying coefficients g_{release} and g_{sb} , it is possible to adjust the relative levels of the string sound, the soundboard response, and the release sound.

In the following, we describe the string model, the sample databases, and the soundboard model in detail, and discuss the need for modeling the dispersion of harpsichord strings.

3.1. Basic string model revisited

We use a version of the vibrating string filter model proposed by Jaffe and Smith [16]. It consists of a feedback loop, where a delay line, a fractional delay filter, a high-order allpass filter, and a loss filter are cascaded. The delay line and the fractional delay filter determine the fundamental frequency of the tone. The high-order allpass filter [16] simulates dispersion which

²The tuning is considerably lower than the current standard (440 Hz or higher). This is typical of old musical instruments.

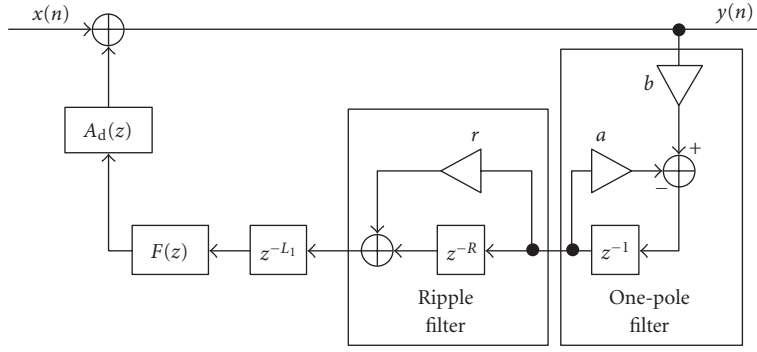


FIGURE 3: Structure of the proposed string model. The feedback loop contains a one-pole filter (denominator of (1)), a feedforward comb filter called “ripple filter” (numerator of (1)), the rest of the delay line, a fractional delay filter $F(z)$, and an allpass filter $A_d(z)$ simulating dispersion.

is a typical characteristic of vibrating strings and which introduces inharmonicity in the sound. For the fractional delay filter, we use a first-order allpass filter, as originally suggested by Smith and Jaffe [16, 17]. This choice was made because it allows a simple and sufficient approximation of delay when a high sampling rate is used.³ Furthermore, there is no need to implement fundamental frequency variations (pitch bend) in harpsichord tones. Thus, the recursive nature of the allpass fractional delay filter, which can cause transients during pitch bends, is not harmful.

The loss filter of waveguide string models is usually implemented as a one-pole filter [18], but now we use an extended version. The transfer function of the new loss filter is

$$H(z) = b \frac{r + z^{-R}}{1 + az^{-1}}, \quad (1)$$

where the scaling parameter b is defined as

$$b = g(1 + a), \quad (2)$$

R is the delay line length of the ripple filter, r is the ripple depth, and a is the feedback gain. Figure 3 shows the block diagram of the string model with details of the new loss filter, which is seen to be composed of the conventional one-pole filter and a ripple filter in cascade. The total delay line length L in the feedback loop is $1 + R + L_1$ plus the phase delay caused by the fractional delay filter $F(z)$ and the allpass filter $A_d(z)$.

The overall loop gain is determined by parameter g , which is usually selected to be slightly smaller than 1 to ensure stability of the feedback loop. The feedback gain parameter a defines the overall lowpass character of the filter: a value slightly smaller than 0 (e.g., $a = -0.01$) yields a mild lowpass filter, which causes high-frequency partials to decay faster than the low-frequency ones, which is natural.

The ripple depth parameter r is used to control the deviation of the loss filter gain from that of the one-pole filter.

The delay line length R is determined as

$$R = \text{round}(r_{\text{rate}}L), \quad (3)$$

where r_{rate} is the ripple rate parameter that adjusts the ripple density in the frequency domain and L is the total delay length in the loop (in samples, or sampling intervals).

The ripple filter was developed because it was found that the magnitude response of the one-pole filter alone is overly smooth when compared to the required loop gain behavior for harpsichord sounds. Note that the ripple factor r in (1) increases the loop gain, but it is not accounted for in the scaling factor in (2). This is purposeful because we find it useful that the loop gain oscillates symmetrically around the magnitude response of the conventional one-pole filter (obtained from (1) by setting $r = 0$). Nevertheless, it must be ensured somehow that the overall loop gain does not exceed unity at any of the harmonic frequencies—otherwise the system becomes unstable. It is sufficient to require that the sum $g + |r|$ remains below one, or $|r| < 1 - g$. In practice, a slightly larger magnitude of r still results in a stable system when $r < 0$, because this choice decreases the loop gain at 0 Hz and the conventional loop filter is a lowpass filter, and thus its gain at the harmonic frequencies is smaller than g .

With small positive or negative values of r , it is possible to obtain wavy loop gain characteristics, where two neighboring partials have considerably different loop gains and thus decay rates. The frequency of the ripple is controlled by parameter r_{rate} so that a value close to one results in a very slow wave, while a value close to 0.5 results in a fast variation where the loop gain for neighboring even and odd partials differs by about $2r$ (depending on the value of a). An example is shown in Figure 4 where the properties of a conventional one-pole loss filter are compared against the proposed ripply loss filter. Figure 4a shows that by adding a feedforward path with small gain factor $r = 0.002$, the loop gain characteristics can be made less regular.

Figure 4b shows the corresponding reverberation time (T_{60}) curve, which indicates how long it takes for each partial to decay by 60 dB. The T_{60} values are obtained by multiplying the time-constant values τ by $-60/[20 \log(1/e)]$ or 6.9078.

³The sampling rate used in this work is 44100 Hz.

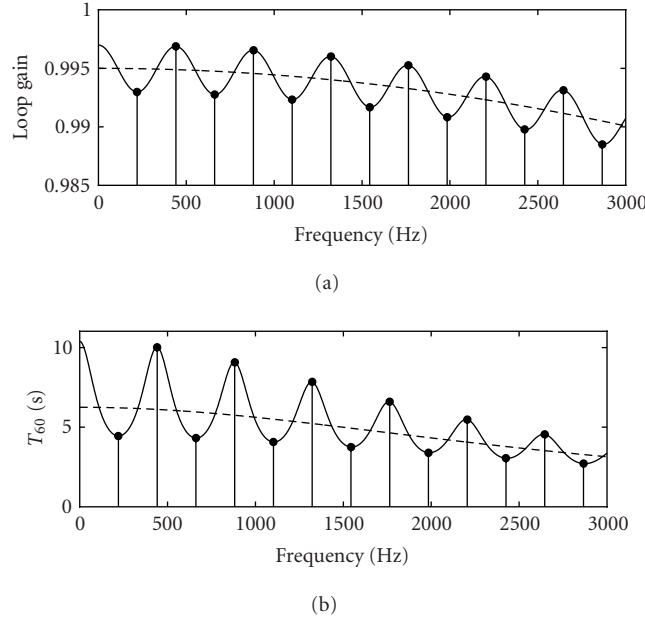


FIGURE 4: The frequency-dependent (a) loop gain (magnitude response) and (b) reverberation time T_{60} determined by the loss filter. The dashed lines show the smooth characteristics of a conventional one-pole loss filter ($g = 0.995$, $a = -0.05$). The solid lines show the characteristics obtained with the ripply loss filter ($g = 0.995$, $a = -0.05$, $r = 0.0020$, $r_{\text{rate}} = 0.5$). The bold dots indicate the actual properties experienced by the partials of the synthetic tone ($L = 200$ samples, $f_0 = 220.5$ Hz).

The time constants $\tau(k)$ for partial indices $k = 1, 2, 3, \dots$, on the other hand, are obtained from the loop gain data $G(k)$ as

$$\tau(k) = \frac{-1}{f_0 \ln[G(k)]}. \quad (4)$$

The loop gain sequence $G(k)$ is extracted directly from the magnitude response of the loop filter at the fundamental frequency ($k = 1$) and at the other partial frequencies ($k = 2, 3, 4, \dots$).

Figure 4b demonstrates the power of the ripply loss filter: the second partial can be rendered to decay much slower than the first and the third partials. This is also perceived in the synthetic tone: soon after the attack, the second partial stands out as the loudest and the longest ringing partial. Formerly, this kind of flexibility has been obtained only with high-order loss filters [17, 19]. Still, the new filter has only two parameters more than the one-pole filter, and its computational complexity is comparable to that of a first-order pole-zero filter.

3.2. Inharmonicity

Dispersion is always present in real strings. It is caused by the stiffness of the string material. This property of strings gives rise to inharmonicity in the sound. An offspring of the harpsichord, the piano, is famous for its strongly inharmonic tones, especially in the bass range [9, 20]. This is due to the large elastic modulus and the large diameter of high-strength steel strings in the piano [9]. In waveguide models, inharmonicity is modeled with allpass filters [16, 21, 22, 23]. Naturally, it would be cost-efficient not to implement the inhar-

monicity, because then the allpass filter $A_d(z)$ would not be needed at all.

The inharmonicity of the recorded harpsichord tones were investigated in order to find out whether it is relevant to model this property. The partials of recorded harpsichord tones were picked semiautomatically from the magnitude spectrum, and with a least-square fit we estimated the inharmonicity coefficient B [20] for each recorded tone. The measured B values are displayed in Figure 5 together with the threshold of audibility and its 90% confidence intervals taken from listening test results [24]. It is seen that the B coefficient is above the mean threshold of audibility in all cases, but above the frequency 140 Hz, the measured values are within the confidence interval. Thus, it is not guaranteed that these cases actually correspond to audible inharmonicity. At low frequencies, in the case of the 19 lowest keys of the harpsichord, where the inharmonicity coefficients are about 10^{-5} , the inharmonicity is audible according to this comparison. It is thus important to implement the inharmonicity for the lowest 2 octaves or so, but it may also be necessary to implement the inharmonicity for the rest of the notes.

This conclusion is in accordance with [10], where inharmonicity is stated as part of the tonal quality of the harpsichord, and also with [12], where it is mentioned that the inharmonicity is less pronounced than in the piano.

3.3. Sample databases

The excitation signals of the string models are stored in a database from where they can be retrieved at the onset time. The excitation sequences contain 20,000 samples (0.45 s),

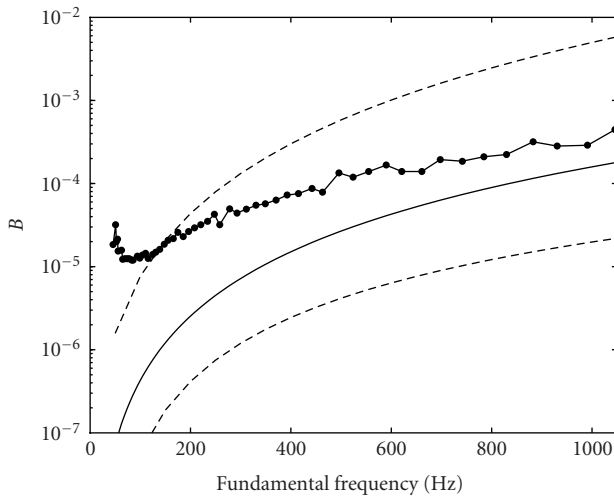


FIGURE 5: Estimates of the inharmonicity coefficient B for all 56 keys of the harpsichord (circles connected with thick line). Also shown are the threshold of audibility for the B coefficient (solid line) and its 90% confidence intervals (dashed lines) taken from [24].

and they have been extracted from recorded tones by canceling the partials. The analysis and calibration procedure is discussed further in Section 4 of this paper. The idea is to include in these samples the sound of the quill scraping the string plus the beginning of the attack of the sound so that a natural attack is obtained during synthesis, and the initial levels of partials are set properly. Note that this approach is slightly different from the standard commuted synthesis technique, where the full inverse filtered recorded signal is used to excite the string model [18, 25]. In the latter case, all modes of the soundboard (or soundbox) are contained within the input sequence, and virtually perfect resynthesis is accomplished if the same parameters are used for inverse filtering and synthesis. In the current model, however, we have truncated the excitation signals by windowing them with the right half of a Hanning window. The soundboard response is much longer than that (several seconds), but imitating its ringing tail is taken care of by the soundboard filter (see the next subsection).

In addition to the excitation samples, we have extracted short release sounds from recorded tones. One of these is retrieved and played each time a note-off command occurs. Extracting these samples is easy: once a note is played, the player can wait until the string sound has completely decayed, and then release the key. This way a clean recording of noises related to the release event is obtained, and any extra processing is unnecessary. An alternative way would be to synthesize these knocking sounds using modal synthesis, as suggested in [26].

3.4. Modeling the reverberant soundboard and undamped strings

When a note is plucked on the harpsichord, the string vibrations excite the bridge and, consequently, the soundboard.

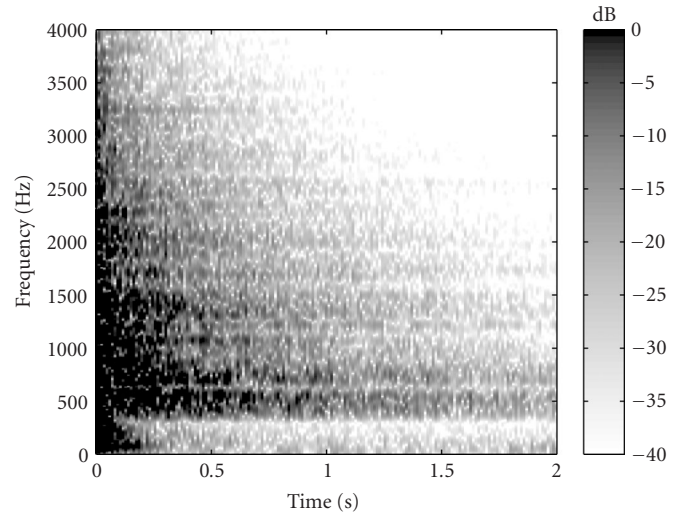


FIGURE 6: Time-frequency plot of the harpsichord air radiation when the 8' bridge is excited. To exemplify the fast decay of the low-frequency modes only the first 2 seconds and frequencies up to 4000 Hz are displayed.

The soundboard has its own modes depending on the size and the materials used. The radiated acoustic response of the harpsichord is reasonably flat over a frequency range from 50 to 2000 Hz [11]. In addition to exciting the air and structural modes of the instrument body, the pluck excites the part of the string that lies behind the bridge, the high modes of the low strings that the dampers cannot perfectly attenuate, and the highest octave of the 4' register strings.⁴ The resonance strings behind the bridge are about 6 to 20 cm long and have a very inharmonic spectral structure. The soundboard filter used in our harpsichord synthesizer (see Figure 2) is responsible for imitating all these features. However, as will be discussed further in Section 4.5, the lowest body modes can be ignored since they decay fast and are present in the excitation samples. In other words, the modeling is divided into two parts so that the soundboard filter models the reverberant tail while the attack part is included in the excitation signal, which is fed to the string model. Reference [11] discusses the resonance modes of the harpsichord soundboard in detail.

The radiated acoustic response of the harpsichord was recorded in an anechoic chamber by exciting the bridges (8' and 4') with an impulse hammer at multiple positions. Figure 6 displays a time-frequency response of the 8' bridge when excited between the C₃ strings, that is, approximately at the middle point of the bridge. The decay times at frequencies below 350 Hz are considerably shorter than in the frequency range from 350 to 1000 Hz. The T_{60} values at the respective bands are about 0.5 seconds and 4.5 seconds. This can be explained by the fact that the short string portions

⁴The instrument used in this study does not have dampers in the last octave of the 4' register.

behind the bridge and the undamped strings resonate and decay slowly.

As suggested by several authors, see for example, [14, 27, 28], the impulse response of a musical instrument body can be modeled with a reverberation algorithm. Such algorithms have been originally devised for imitating the impulse response of concert halls. In a previous work, we triggered a static sample of the body response with every note [29]. In contrast to the sample-based solution, which produces the same response every time, the reverberation algorithm produces additional variation in the sound: as the input signal of the reverberation algorithm is changed, or in this case as the key or register is changed, the temporal and frequency content of the output changes accordingly.

The soundboard response of the harpsichord in this work is modeled with an algorithm presented in [30]. It is a modification of the feedback delay network [31], where the feedback matrix is replaced with a single coefficient, and comb allpass filters have been inserted in the delay line loops. A schematic view of the reverberation algorithm is shown in Figure 7. This structure is used because of its computational efficiency. The $H_k(z)$ blocks represent the loss filters, $A_k(z)$ blocks are the comb allpass filters, and the delay lines are of length P_k . In this work, eight ($N = 8$) delay lines are implemented.

One-pole lowpass filters are used as loss filters which implement the frequency-dependent decay. The comb allpass filters increase the diffusion effect and they all have the transfer function

$$A_k(z) = \frac{a_{ap,k} + z^{-M_k}}{1 + a_{ap,k}z^{-M_k}}, \quad (5)$$

where M_k are the delay-line lengths and $a_{ap,k}$ are the allpass filter coefficients. To ensure stability, it is required that $a_{ap,k} \in [-1, 1]$. In addition to the reverberation algorithm, a tone-corrector filter, as shown in Figure 2, is used to match the spectral envelope of the target response, that is, to suppress the low frequencies below 350 Hz and give some additional lowpass characteristics at high frequencies. The choice of the parameters is discussed in Section 4.5.

4. CALIBRATION OF THE SYNTHESIS ALGORITHM

The harpsichord was brought into an anechoic chamber where the recordings and the acoustic measurements were conducted. The registered signals enable the automatic calibration of the harpsichord synthesizer. This section describes the recordings, the signal analysis, and the calibration techniques for the string and the soundboard models.

4.1. Recordings

Harpsichord tones were recorded in the large anechoic chamber of Helsinki University of Technology. Recordings were made with multiple microphones installed at a distance of about 1 m above the soundboard. The signals were recorded digitally (44.1 kHz, 16 bits) directly onto the hard disk, and to remove disturbances in the infrasonic range, they were

highpass filtered. The highpass filter is a fourth-order Butterworth highpass filter with a cutoff frequency of 52 Hz or 32 Hz (for the lowest tones). The filter was applied to the signal in both directions to obtain a zero-phase filtering. The recordings were compared in an informal listening test among the authors, and the signals obtained with a high-quality studio microphone by Schoeps were selected for further analysis.

All 56 keys of the instrument were played separately with six different combinations of the registers that are commonly used. This resulted in $56 \times 6 = 336$ recordings. The tones were allowed to decay into silence, and the key release was included. The length of the single tones varied between 10 and 25 seconds, because the bass tones of the harpsichord tend to ring much longer than the treble tones. For completeness, we recorded examples of different dynamic levels of different keys, although it is known that the harpsichord has a limited dynamic range due to its excitation mechanism. Short staccato tones, slow key pressings, and fast repetitions of single keys were also registered. Chords were recorded to measure the variations of attack times between simultaneously played keys. Additionally, scales and excerpts of musical pieces were played and recorded.

Both bridges of the instrument were excited at several points (four and six points for the 4' and the 8' bridge, respectively) with an impulse hammer to obtain reliable acoustic soundboard responses. The force signal of the hammer and acceleration signal obtained from an accelerometer attached to the bridge were recorded for the 8' bridge at three locations. The acoustic response was recorded in synchrony.

4.2. Analysis of recorded tones and extraction of excitation signals

Initial estimates of the synthesizer parameters can be obtained from analysis of recorded tones. For the basic calibration of the synthesizer, the recordings were selected where each register is played alone. We use a method based on the short-time Fourier transform and sinusoidal modeling, as previously discussed in [18, 32]. The inharmonicity of harpsichord tones is accounted for in the spectral peak-picking algorithm with the help of the estimated B coefficient values. After extracting the fundamental frequency, the analysis system essentially decomposes the analyzed tone into its deterministic and stochastic parts, as in the spectral modeling synthesis method [33]. However, in our system the decay times of the partials are extracted, and the loop filter design is based on the loop gain data calculated from the decay times. The envelopes of partials in the harpsichord tones exhibit beating and two-stage decay, as is usual for string instruments [34]. The residual is further processed, that is, the soundboard contribution is mostly removed (by windowing the residual signal in the time domain) and the initial level of each partial is adjusted by adding a correction obtained through sinusoidal modeling and inverse filtering [35, 36]. The resulting processed residual is used as an excitation signal to the model.

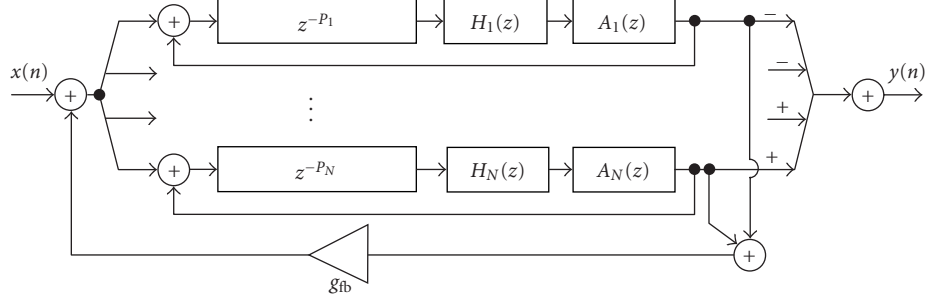


FIGURE 7: A schematic view of the reverberation algorithm used for soundboard modeling.

4.3. Loss filter design

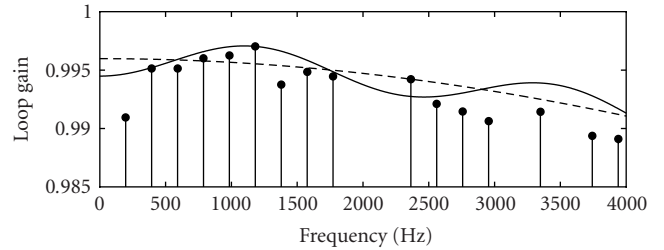
Since the ripply loop filter is an extension of the one-pole filter that allows improved matching of the decay rate of one partial and simply introduces variations to the others, it is reasonable to design it after the one-pole filter. This kind of approach is known to be suboptimal in filter design, but highest possible accuracy is not the main goal of this work. Rather, a simple and reliable routine to automatically process a large amount of measurement data is reached for, thus leaving a minimum amount of erroneous results to be fixed manually.

Figure 8 shows the loop gain and T_{60} data for an example case. It is seen that the target data (bold dots in Figure 8) contain a fair amount of variation from one partial to the next one, although the overall trend is downward as a function of frequency. Partial indices 10, 11, 16, and 18 are excluded (set to zero), because their decay times were found to be unreliable (i.e., loop gain larger than unity). The one-pole filter response fitted using a weighted least squares technique [18] (dashed lines in Figure 8) can follow the overall trend, but it evens up the differences between neighboring partials.

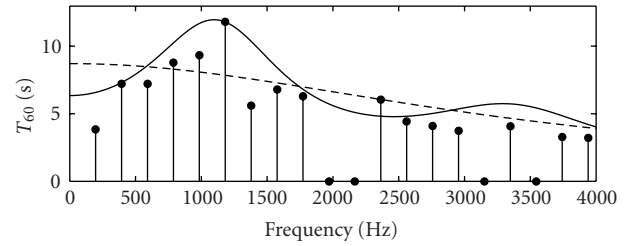
The ripply loss filter can be designed using the following heuristic rules.

- (1) Select the partial with the largest loop gain starting from the second partial⁵ (the sixth partial in this case, see Figure 8), whose index is denoted by k_{\max} . Usually one of the lowest partials will be picked once the outliers have been discarded.
- (2) Set the absolute value of r so that, together with the one-pole filter, the magnitude response will match the target loop gain of the partial with index k_{\max} , that is, $|r| = G(k_{\max}) - |H(k_{\max}, f_0)|$, where the second term is the loop gain due to the one-pole filter at that frequency (in this case $r = 0.0015$).

⁵In practice, the first partial may have the largest loop gain. However, if we tried to match it using the ripply loss filter, the r_{rate} parameter would go to 1, as can be seen from (6), and the delay-line length R would become equal to L rounded to an integer, as can be seen from (3). This practically means that the ripple filter would be reduced to a correction of the loop gain by r , which can be done also by simply replacing the loop gain parameter g by $g + r$. For this reason, it is sensible to match the loop gain of a partial other than the first one.



(a)



(b)

FIGURE 8: (a) The target loop gain for a harpsichord tone ($f_0 = 197$ Hz) (bold dots), the magnitude response of the conventional one-pole filter with $g = 0.9960$ and $a = -0.0296$ (dashed line), and the magnitude response of the ripply loss filter with $r = -0.0015$ and $r_{\text{rate}} = 0.0833$ (solid line). (b) The corresponding T_{60} data. The total delay-line length is 223.9 samples, and the delay-line length R of the ripple filter is 19 samples.

- (3) If the target loop gain of the first partial is larger than the magnitude response of the one-pole filter alone at that frequency, set the sign of r to positive, and otherwise to negative so that the decay of the first partial is made fast (in the example case in Figure 8, the minus sign is chosen, that is, $r = -0.0015$).
- (4) If a positive r has been chosen, conduct a stability check at the zero frequency. If it fails (i.e., $g + r \geq 1$), the value of r must be made negative by changing its sign.
- (5) Set the ripple rate parameter r_{rate} so that the longest ringing partial will occur at the maximum nearest to 0 Hz. This means that the parameter must be chosen

according to the following rule:

$$r_{\text{rate}} = \begin{cases} \frac{1}{k_{\text{max}}} & \text{when } r \geq 0, \\ \frac{1}{2k_{\text{max}}} & \text{when } r < 0. \end{cases} \quad (6)$$

In the example case, as the ripple pattern is a negative cosine wave (in the frequency domain) and the peak should hit the 6th partial, we set the r_{rate} parameter equal to $1/12 = 0.0833$. This implies that the minimum will occur at every 12th partial and the first maximum will occur at the 6th partial. The result of this design procedure is shown in Figure 8 with the solid line. Note that the peak is actually between the 5th and the 6th partial, because fractional delay techniques are not used in this part of the system and the delay-line length R is thus an integer, as defined in (3). It is obvious that this design method is limited in its ability to follow arbitrary target data. However, as we now know that the resolution of human hearing is also very limited in evaluating differences in decay rates [37], we find the match in most cases to be sufficiently good.

4.4. Beating filter design

The beating filter, a second-order resonator $R(z)$ coupled in parallel with the string model (see Figure 2), is used for reproducing the beating in harpsichord synthesis. In practice, we decided to choose the center frequency of the resonator so that it brings about the beating effect in one of the low-index partials that has a prominent level and large beat amplitude. These criteria make sure that the single resonator will produce an audible effect during synthesis.

In this implementation, we probed the deviation of the actual decay characteristics of the partials from the ideal exponential decay. This procedure is illustrated in Figure 9. In Figure 9a, the mean-squared error (MSE) of the deviation is shown. The lowest partial that exhibits a high deviation (10th partial in this example) is selected as a candidate for the most prominent beating partial. Its magnitude envelope is presented in Figure 9b by a solid curve. It exhibits a slow beating pattern with a period of about 1.5 seconds. The second-order resonator that simulates beating, in turn, can be tuned to result in a beating pattern with this same rate. For comparison, the magnitude envelopes of the 9th and 11th partials are also shown by dashed and dash-dotted curves, respectively.

The center frequency of the resonator is measured from the envelope of the partial. In practice, the offset ranges from practically 0 Hz to a few Hertz. The gain of the resonator, that is, the amplitude of the beating partial, is set to be the same as that of the partial it beats against. This simple choice is backed by the recent result by Järveläinen and Karjalainen [38] that the beating in string instrument tones is essentially perceived as an on/off process: if the beating amplitude is above the threshold of audibility, it is noticed, while if it is below it, it becomes inaudible. Furthermore, changes in the beating amplitude appear to be inaccurately perceived. Before knowing these results, in a former version of the synthesizer, we also decided to use the same amplitude for the two

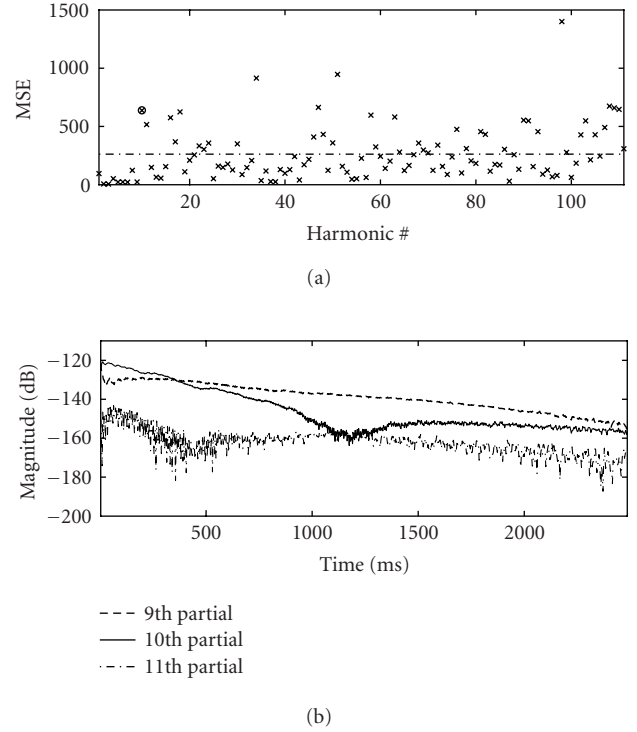


FIGURE 9: (a) The mean squared error of exponential curve fitting to the decay of partials ($f_0 = 197$ Hz), where the lowest large deviation has been circled (10th partial), and the acceptance threshold is presented with a dashed-dotted line. (b) The corresponding temporal envelopes of the 9th, 10th, and 11th partials, where the slow beating of the 10th partial and deviations in decay rates are visible.

components that produce the beating, because the mixing parameter that adjusts the beating amplitude was not giving a useful audible variation [39]. Thus, we are now convinced that it is unnecessary to add another parameter for all string models by allowing changes in the amplitude of the beating partial.

4.5. Design of soundboard filter

The reverberation algorithm and the tone correction unit are set in cascade and together they form the soundboard model, as shown in Figure 2. For determining the soundboard filter, the parameters of the reverberation algorithm and its tone corrector have to be set. The parameters for the reverberation algorithm were chosen as proposed in [31]. To match the frequency-dependent decay, the ratio between the decay times at 0 Hz and at $f_s/2$ was set to 0.13, so that T_{60} at 0 Hz became 6.0 seconds. The lengths of the eight delay lines varied from 1009 to 1999 samples. To avoid superimposing the responses, the lengths were incommensurate numbers [40]. The lengths M_k of the delay lines in the comb allpass structures were set to 8% of the total length of each delay line path P_k , filter coefficients $a_{\text{ap},k}$ were all set to 0.5, and the feedback coefficient g_{fb} was set to -0.25 .

The excitation signals for the harpsichord synthesizer are 0.45 second long, and hence contain the necessary fast-decaying modes for frequencies below 350 Hz (see Figure 6). Therefore, the tone correction section is divided into two parts: a highpass filter that suppresses frequencies below 350 Hz and another filter that imitates the spectral envelope at the middle and high frequencies. The highpass filter is a 5th-order Chebyshev type I design with a 5 dB passband ripple, the 6 dB point at 350 Hz, and a roll-off rate of about 50 dB per octave below the cutoff frequency. The spectral envelope filter for the soundboard model is a 10th-order IIR filter designed using linear prediction [41] from a 0.2-second long windowed segment of the measured target response (see Figure 6 from 0.3 second to 0.5 second). Figure 10 shows the time-frequency plot of the target response and the soundboard filter for the first 1.5 seconds up to 10 kHz. The target response has a prominent lowpass characteristic, which is due to the properties of the impulse hammer. While the response should really be inverse filtered by the hammer force signal, in practice we can approximately compensate this effect with a differentiator whose transfer function is $H_{\text{diff}}(z) = 0.5 - 0.5z^{-1}$. This is done before the design of the tone corrector, so the compensation filter is not included in the synthesizer implementation.

5. IMPLEMENTATION AND APPLICATIONS

This section deals with computational efficiency, implementation issues, and musical applications of the harpsichord synthesizer.

5.1. Computational complexity

The computational cost caused by implementing the harpsichord synthesizer and running it at an audio sample rate, such as 44100 Hz, is relatively small. Table 1 summarizes the amount of multiplications and additions needed per sample for various parts of the system. In this cost analysis, it is assumed that the dispersion is simulated using a first-order allpass filter. In practice, the lowest tones require a higher-order allpass filter, but some of the highest tones may not have the allpass filter at all. So the first-order filter represents an average cost per string model. Note that the total cost per string is smaller than that of an FIR filter of order 12 (i.e., 13 multiplications and 12 additions). In practice, one voice in harpsichord synthesis is allocated one to three string models, which simulate the different registers. The soundboard model is considerably more costly than a string model: the number of multiplications is more than fourfold, and the number of additions is almost seven times larger. The complexity analysis of the comb allpass filters in the soundboard model is based on the direct form II implementation (i.e., one delay line, two multiplications, and two additions per comb allpass filter section).

The implementation of the synthesizer, which is discussed in detail in the next section, is based on high-level programming and control. Thus, it is not optimized for fastest possible real-time operation. The current implementation of the synthesizer runs on a Macintosh G4 (800 MHz)

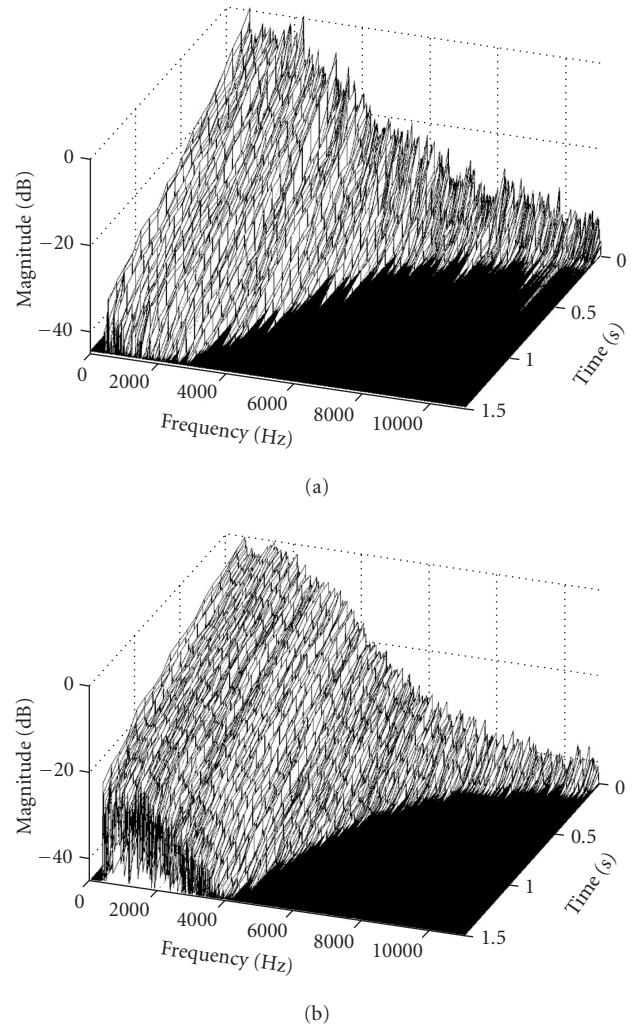


FIGURE 10: The time-frequency representation of (a) the recorded soundboard response and (b) the synthetic response obtained as the impulse response of a modified feedback delay network.

computer, and it can simultaneously run 15 string models in real time without the soundboard model. With the soundboard model, it is possible to run about 10 strings. A new, faster computer and optimization of the code can increase these numbers. With optimized code and fast hardware, it may be possible to run the harpsichord synthesizer with full polyphony (i.e., 56 voices) and soundboard in real time using current technology.

5.2. Synthesizer implementation

The signal-processing part of the harpsichord synthesizer is realized using a visual software synthesis package called PWSynth [42]. PWSynth, in turn, is part of a larger visual programming environment called PWGL [43]. Finally, the control information is generated using our music notation package ENP (expressive notation package) [44]. In this section, the focus is on design issues that we have encountered when implementing the synthesizer. We also give ideas on

TABLE 1: The number of multiplications and additions in different parts of the synthesizer.

Part of synthesis algorithm	Multiplications	Additions
String model		
• Fractional delay allpass filter $F(z)$	2	2
• Inharmonizing allpass filter $A_d(z)$	2	2
• One-pole filter	2	1
• Ripple filter	1	1
• Resonator $R(z)$	3	2
• Timbre control	2	1
• Mixing with release sample	1	1
Soundboard model		
• Modified FDN reverberator	33	47
• IIR tone corrector	11	10
• Highpass filter	12	9
• Mixing	1	1
Total		
• Per string (without soundboard model)	13	10
• Soundboard model	57	67
• All (one string and soundboard model)	70	77

how the model is parameterized so that it can be controlled from the music notation software.

Our previous work in designing computer simulations of musical instruments has resulted in several applications, such as the classical guitar [39], the Renaissance lute, the Turkish ud [45], and the clavichord [29]. The two-manual harpsichord tackled in the current study is the most challenging and complex instrument that we have yet investigated. As this kind of work is experimental, and the synthesis model must be refined by interactive listening, a system is needed that is capable of making fast and efficient prototypes of the basic components of the system. Another non-trivial problem is the parameterization of the harpsichord synthesizer. In a typical case, one basic component, such as the vibrating string model, requires over 10 parameters so that it can be used in a convincing simulation. Thus, since the full harpsichord synthesizer implementation has three string sets each having 56 strings, we need at least 1680 ($= 10 \times 3 \times 56$) parameters in order to control all individual strings separately.

Figure 11 shows a prototype of a harpsichord synthesizer. It contains three main parts. First, the top-most box (called “num-box” with the label “number-of-strings”) gives the number of strings within each string set used by the synthesizer. This number can vary from 1 (useful for preliminary tests) to 56 (the full instrument). In a typical real-time situation, this number can vary, depending on the polyphony of the musical score to be realized, between 4 and 10. The next box of interest is called “string model.” It is a special abstraction box that contains a subwindow. The contents of this window are displayed in Figure 12. This abstraction box defines a single string model. Next, Figure 11 shows three “copy-synth-patch” boxes that determine the individ-

ual string sets used by the instrument. These sets are labeled as follows: “harpsy1/8-fb/,” “harpsy1/8-ff,” and “harpsy1/4-ff/.” Each string set copies the string model patch *count* times, where *count* is equal to the current number of strings (given by the upper number-of-strings box). The rest of the boxes in the patch are used to mix the outputs of the string sets.

Figure 12 gives the definition of a single string model. The patch consists of two types of boxes. First, the boxes with the name “pwsynth-plug” (the boxes with the darkest outlines in grey-scale) define the parametric entry points that are used by our control system. Second, the other boxes are low-level DSP modules, realized in C++, that perform the actual sample calculation and boxes which are used to initialize the DSP modules. The “pwsynth-plug” boxes point to memory addresses that are continuously updated while the synthesizer is running. Each “pwsynth-plug” box has a label that is used to build symbolic parameter pathnames. While the “copy-synth-patch” boxes (see the main patch of Figure 11) copy the string model in a loop, the system automatically generates new unique pathnames by merging the label from the current “copy-synth-patch” box, the current loop index, and the label found in “pwsynth-plug” boxes. Thus, pathnames like “harpsy1/8-fb/1/lfgain” are obtained, which refers to the lfgain (loss filter gain) of the first string of the 8’ back string set of a harpsichord model called “harpsy1.”

5.3. Musical applications

The harpsichord synthesizer can be used as an electronic musical instrument controlled either from a MIDI keyboard or from a sequencer software. Recently, some composers have been interested in using a formerly developed model-based guitar synthesizer for compositions, which are either experimental in nature or extremely challenging for human players.

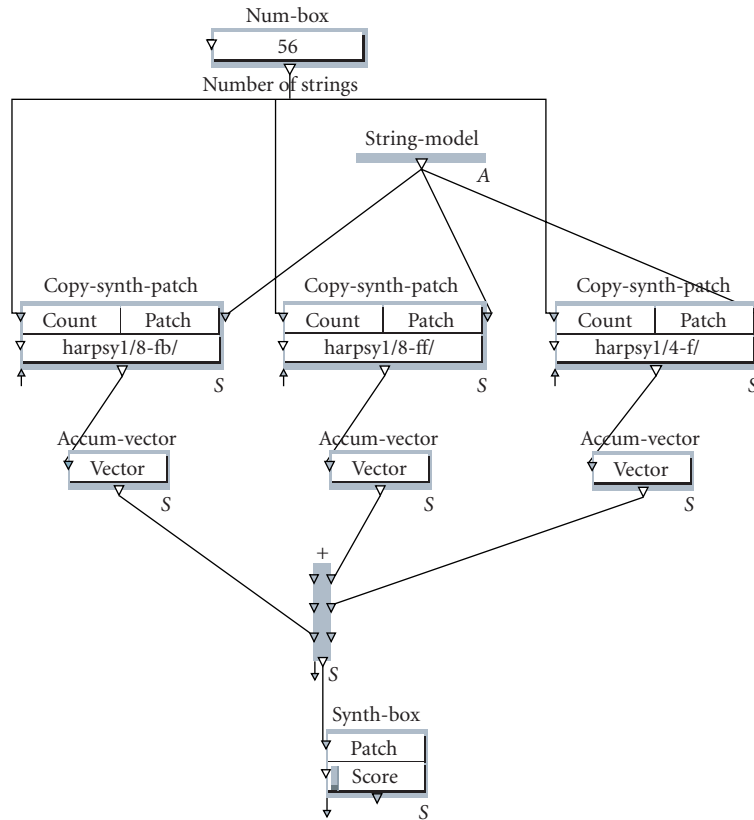


FIGURE 11: The top-level prototype of the harpsichord synthesizer in PWSynth. The patch defines one string model and the three string sets used by the instrument.

Another fascinating idea is to extend the range and timbre of the instrument. A version of the guitar synthesizer, that we call the super guitar, has an extended range and a large number of strings [46]. We plan to develop a similar extension of the harpsichord synthesizer.

In the current version of the synthesizer, the parameters have been calibrated based on recordings. One obvious application for a parametric synthesizer is to modify the timbre by deviating the parameter values. This can lead to extended timbres that belong to the same instrument family as the original instrument or, in the extreme cases, to a novel virtual instrument that cannot be recognized by listeners. One of the most obvious subjects for modification is the decay rate, which is controlled with the coefficients of the loop filter.

A well-known limitation of the harpsichord is its restricted dynamic range. In fact, it is a controversial issue whether the key velocity has any audible effect on the sound of the harpsichord. The synthesizer easily allows the implementation of an exaggerated dynamic control, where the key velocity has a dramatic effect on both the amplitude and the timbre, if desired, such as in the piano or in the acoustic guitar. As the key velocity information is readily available, it can be used to control the gain and the properties of a timbre control filter (see Figure 2).

Luthiers who make musical instruments are interested in

modern technology and want to try physics-based synthesis to learn about the instrument. A synthesizer allows varying certain parameters in the instrument design, which are difficult or impossible to adjust in the real instrument. For example, the point where the quill plucks the string is structurally fixed in the harpsichord, but as it has a clear effect on the timbre, varying it is of interest. In the current harpsichord synthesizer, it would require the knowledge of the plucking point and then inverse filtering its contribution from the excitation signal. The plucking point contribution can then be implemented in the string model by inserting another feed-forward comb filter, as discussed previously in several works [7, 16, 17, 18]. Another prospect is to vary the location of the damper. Currently, we do not have an exact model for the damper, and neither is its location a parameter. Testing this is still possible, because it is known that the nonideal functioning of the damper is related to the nodal points of the strings, which coincide with the locations of the damper. The ripply loss filter allows the imitation of this effect.

Luthiers are interested in the possibility of virtual prototyping without the need for actually building many versions of an instrument out of wood. The current synthesis model may not be sufficiently detailed for this purpose. A real-time or near-real-time implementation of a physical model, where several parameters can be adjusted, would be an ideal tool for testing prototypes.

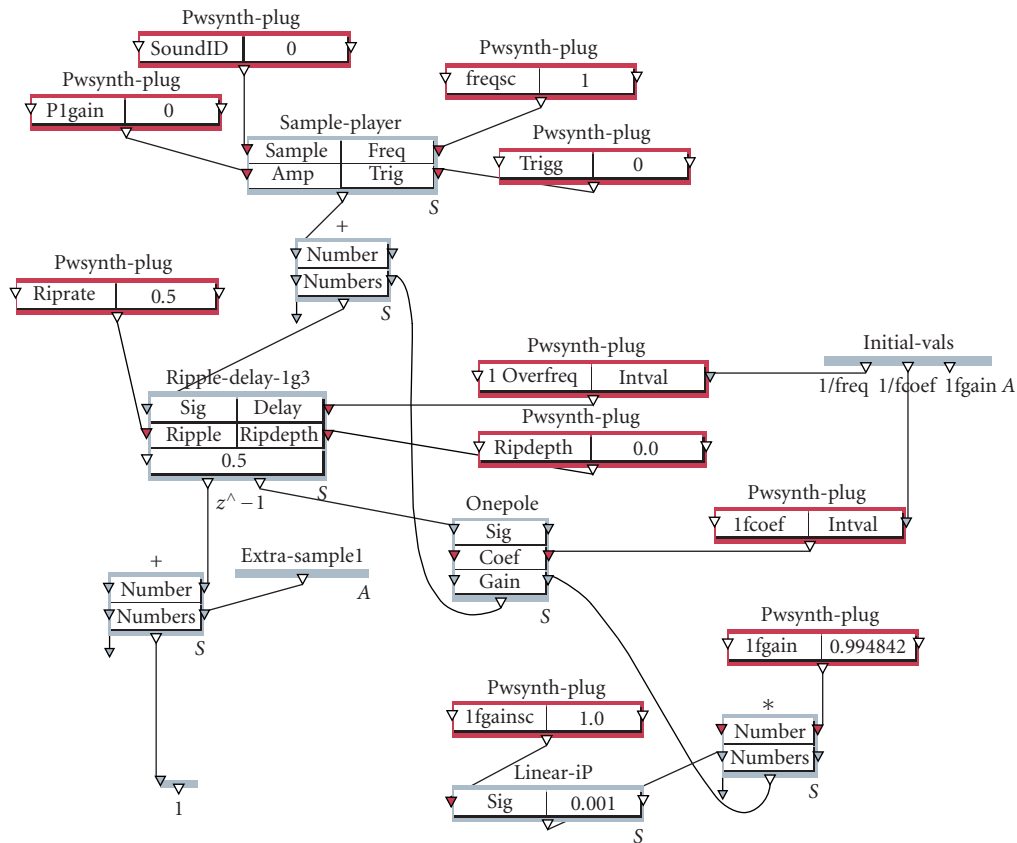


FIGURE 12: The string model patch. The patch contains the low-level DSP modules and parameter entry points used by the harpsichord synthesizer.

6. CONCLUSIONS

This paper proposes signal-processing techniques for synthesizing harpsichord tones. A new extension to the loss filter of the waveguide synthesizer has been developed which allows variations in the decay times of neighboring partials. This filter will be useful also for the waveguide synthesis of other stringed instruments. The fast-decaying modes of the soundboard are incorporated in the excitation samples of the synthesizer, while the long-ringing modes at the middle and high frequencies are imitated using a reverberation algorithm. The calibration of the synthesis model is made almost automatic. The parameterization and use of simple filters also allow manual adjustment of the timbre. A physics-based synthesizer, such as the one described here, has several musical applications, the most obvious one being the usage as a computer-controlled musical instrument.

Examples of single tones and musical pieces synthesized with the synthesizer are available at <http://www.acoustics.hut.fi/publications/papers/jasp-harpsy/>.

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REFERENCES

- [1] J. O. Smith, "Efficient synthesis of stringed musical instruments," in *Proc. International Computer Music Conference*, pp. 64–71, Tokyo, Japan, September 1993.
- [2] M. Karjalainen and V. Välimäki, "Model-based analysis/synthesis of the acoustic guitar," in *Proc. Stockholm Music Acoustics Conference*, pp. 443–447, Stockholm, Sweden, July–August 1993.
- [3] M. Karjalainen, V. Välimäki, and Z. Jánosy, "Towards high-quality sound synthesis of the guitar and string instruments," in *Proc. International Computer Music Conference*, pp. 56–63, Tokyo, Japan, September 1993.
- [4] J. O. Smith and S. A. Van Duyne, "Commutated piano synthesis," in *Proc. International Computer Music Conference*, pp. 319–326, Banff, Alberta, Canada, September 1995.
- [5] J. O. Smith, "Physical modeling using digital waveguides," *Computer Music Journal*, vol. 16, no. 4, pp. 74–91, 1992.
- [6] K. Karplus and A. Strong, "Digital synthesis of plucked string and drum timbres," *Computer Music Journal*, vol. 7, no. 2, pp. 43–55, 1983.
- [7] M. Karjalainen, V. Välimäki, and T. Tolonen, "Plucked-string models, from the Karplus-Strong algorithm to digital

- waveguides and beyond,” *Computer Music Journal*, vol. 22, no. 3, pp. 17–32, 1998.
- [8] F. Hubbard, *Three Centuries of Harpsichord Making*, Harvard University Press, Cambridge, Mass, USA, 1965.
 - [9] N. H. Fletcher and T. D. Rossing, *The Physics of Musical Instruments*, Springer-Verlag, New York, NY, USA, 1991.
 - [10] E. L. Kottick, K. D. Marshall, and T. J. Hendrickson, “The acoustics of the harpsichord,” *Scientific American*, vol. 264, no. 2, pp. 94–99, 1991.
 - [11] W. R. Savage, E. L. Kottick, T. J. Hendrickson, and K. D. Marshall, “Air and structural modes of a harpsichord,” *Journal of the Acoustical Society of America*, vol. 91, no. 4, pp. 2180–2189, 1992.
 - [12] N. H. Fletcher, “Analysis of the design and performance of harpsichords,” *Acustica*, vol. 37, pp. 139–147, 1977.
 - [13] J. Sankey and W. A. Sethares, “A consonance-based approach to the harpsichord tuning of Domenico Scarlatti,” *Journal of the Acoustical Society of America*, vol. 101, no. 4, pp. 2332–2337, 1997.
 - [14] B. Bank, “Physics-based sound synthesis of the piano,” M.S. thesis, Department of Measurement and Information Systems, Budapest University of Technology and Economics, Budapest, Hungary, 2000, published as Tech. Rep. 54, Laboratory of Acoustics and Audio Signal Processing, Helsinki University of Technology, Espoo, Finland, 2000.
 - [15] B. Bank, V. Välimäki, L. Sujbert, and M. Karjalainen, “Efficient physics based sound synthesis of the piano using DSP methods,” in *Proc. European Signal Processing Conference*, vol. 4, pp. 2225–2228, Tampere, Finland, September 2000.
 - [16] D. A. Jaffe and J. O. Smith, “Extensions of the Karplus-Strong plucked-string algorithm,” *Computer Music Journal*, vol. 7, no. 2, pp. 56–69, 1983.
 - [17] J. O. Smith, *Techniques for digital filter design and system identification with application to the violin*, Ph.D. thesis, Stanford University, Stanford, Calif, USA, 1983.
 - [18] V. Välimäki, J. Huopaniemi, M. Karjalainen, and Z. Jánosy, “Physical modeling of plucked string instruments with application to real-time sound synthesis,” *Journal of the Audio Engineering Society*, vol. 44, no. 5, pp. 331–353, 1996.
 - [19] B. Bank and V. Välimäki, “Robust loss filter design for digital waveguide synthesis of string tones,” *IEEE Signal Processing Letters*, vol. 10, no. 1, pp. 18–20, 2003.
 - [20] H. Fletcher, E. D. Blackham, and R. S. Stratton, “Quality of piano tones,” *Journal of the Acoustical Society of America*, vol. 34, no. 6, pp. 749–761, 1962.
 - [21] S. A. Van Duyne and J. O. Smith, “A simplified approach to modeling dispersion caused by stiffness in strings and plates,” in *Proc. International Computer Music Conference*, pp. 407–410, Århus, Denmark, September 1994.
 - [22] D. Rocchesso and F. Scalcon, “Accurate dispersion simulation for piano strings,” in *Proc. Nordic Acoustical Meeting*, pp. 407–414, Helsinki, Finland, June 1996.
 - [23] B. Bank, F. Avanzini, G. Borin, G. De Poli, F. Fontana, and D. Rocchesso, “Physically informed signal processing methods for piano sound synthesis: a research overview,” *EURASIP Journal on Applied Signal Processing*, vol. 2003, no. 10, pp. 941–952, 2003.
 - [24] H. Järveläinen, V. Välimäki, and M. Karjalainen, “Audibility of the timbral effects of inharmonicity in stringed instrument tones,” *Acoustics Research Letters Online*, vol. 2, no. 3, pp. 79–84, 2001.
 - [25] M. Karjalainen and J. O. Smith, “Body modeling techniques for string instrument synthesis,” in *Proc. International Computer Music Conference*, pp. 232–239, Hong Kong, China, August 1996.
 - [26] P. R. Cook, “Physically informed sonic modeling (PhISM): synthesis of percussive sounds,” *Computer Music Journal*, vol. 21, no. 3, pp. 38–49, 1997.
 - [27] D. Rocchesso, “Multiple feedback delay networks for sound processing,” in *Proc. X Colloquio di Informatica Musicale*, pp. 202–209, Milan, Italy, December 1993.
 - [28] H. Penttinen, M. Karjalainen, T. Paatero, and H. Järveläinen, “New techniques to model reverberant instrument body responses,” in *Proc. International Computer Music Conference*, pp. 182–185, Havana, Cuba, September 2001.
 - [29] V. Välimäki, M. Laurson, and C. Erkut, “Commuted waveguide synthesis of the clavichord,” *Computer Music Journal*, vol. 27, no. 1, pp. 71–82, 2003.
 - [30] R. Väänänen, V. Välimäki, J. Huopaniemi, and M. Karjalainen, “Efficient and parametric reverberator for room acoustics modeling,” in *Proc. International Computer Music Conference*, pp. 200–203, Thessaloniki, Greece, September 1997.
 - [31] J. M. Jot and A. Chaigne, “Digital delay networks for designing artificial reverberators,” in *Proc. 90th Convention Audio Engineering Society*, Paris, France, February 1991.
 - [32] C. Erkut, V. Välimäki, M. Karjalainen, and M. Laurson, “Extraction of physical and expressive parameters for model-based sound synthesis of the classical guitar,” in *Proc. 108th Convention Audio Engineering Society*, p. 17, Paris, France, February 2000.
 - [33] X. Serra and J. O. Smith, “Spectral modeling synthesis: a sound analysis/synthesis system based on a deterministic plus stochastic decomposition,” *Computer Music Journal*, vol. 14, no. 4, pp. 12–24, 1990.
 - [34] G. Weinreich, “Coupled piano strings,” *Journal of the Acoustical Society of America*, vol. 62, no. 6, pp. 1474–1484, 1977.
 - [35] V. Välimäki and T. Tolonen, “Development and calibration of a guitar synthesizer,” *Journal of the Audio Engineering Society*, vol. 46, no. 9, pp. 766–778, 1998.
 - [36] T. Tolonen, “Model-based analysis and resynthesis of acoustic guitar tones,” M.S. thesis, Laboratory of Acoustics and Audio Signal Processing, Department of Electrical and Communications Engineering, Helsinki University of Technology, Espoo, Finland, 1998, Tech. Rep. 46.
 - [37] H. Järveläinen and T. Tolonen, “Perceptual tolerances for decay parameters in plucked string synthesis,” *Journal of the Audio Engineering Society*, vol. 49, no. 11, pp. 1049–1059, 2001.
 - [38] H. Järveläinen and M. Karjalainen, “Perception of beating and two-stage decay in dual-polarization string models,” in *Proc. International Symposium on Musical Acoustics*, Mexico City, Mexico, December 2002.
 - [39] M. Laurson, C. Erkut, V. Välimäki, and M. Kuuskankare, “Methods for modeling realistic playing in acoustic guitar synthesis,” *Computer Music Journal*, vol. 25, no. 3, pp. 38–49, 2001.
 - [40] W. G. Gardner, “Reverberation algorithms,” in *Applications of Digital Signal Processing to Audio and Acoustics*, M. Kahrs and K. Brandenburg, Eds., pp. 85–131, Kluwer Academic, Boston, Mass, USA, 1998.
 - [41] J. D. Markel and A. H. Gray Jr., *Linear Prediction of Speech*, Springer-Verlag, Berlin, Germany, 1976.
 - [42] M. Laurson and M. Kuuskankare, “PWSynth: a Lisp-based bridge between computer assisted composition and sound synthesis,” in *Proc. International Computer Music Conference*, pp. 127–130, Havana, Cuba, September 2001.
 - [43] M. Laurson and M. Kuuskankare, “PWGL: a novel visual language based on Common Lisp, CLOS and OpenGL,” in *Proc. International Computer Music Conference*, pp. 142–145, Gothenburg, Sweden, September 2002.

- [44] M. Kuuskankare and M. Laurson, "ENP2.0: a music notation program implemented in Common Lisp and OpenGL," in *Proc. International Computer Music Conference*, pp. 463–466, Gothenburg, Sweden, September 2002.
- [45] C. Erkut, M. Laurson, M. Kuuskankare, and V. Välimäki, "Model-based synthesis of the ud and the Renaissance lute," in *Proc. International Computer Music Conference*, pp. 119–122, Havana, Cuba, September 2001.
- [46] M. Laurson, V. Välimäki, and C. Erkut, "Production of virtual acoustic guitar music," in *Proc. Audio Engineering Society 22nd International Conference on Virtual, Synthetic and Entertainment Audio*, pp. 249–255, Espoo, Finland, June 2002.

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Special Issue on Dependable Semantic Inference

Call for Papers

After many years of exciting research, the field of multimedia information retrieval (MIR) has become mature enough to enter a new development phase—the phase in which MIR technology is made ready to get adopted in practical solutions and realistic application scenarios. High users' expectations in such scenarios require high dependability of MIR systems. For example, in view of the paradigm “getting the content I like, anytime and anyplace” the service of consumer-oriented MIR solutions (e.g., a PVR, mobile video, music retrieval, web search) will need to be at least as dependable as turning a TV set on and off. Dependability plays even a more critical role in automated surveillance solutions relying on MIR technology to analyze recorded scenes and events and alert the authorities when necessary.

This special issue addresses the dependability of those critical parts of MIR systems dealing with semantic inference. Semantic inference stands for the theories and algorithms designed to relate multimedia data to semantic-level descriptors to allow content-based search, retrieval, and management of data. An increase in semantic inference dependability could be achieved in several ways. For instance, better understanding of the processes underlying semantic concept detection could help forecast, prevent, or correct possible semantic inference errors. Furthermore, the theory of using redundancy for building reliable structures from less reliable components could be applied to integrate “isolated” semantic inference algorithms into a network characterized by distributed and collaborative intelligence (e.g., a social/P2P network) and let them benefit from the processes taking place in such a network (e.g., tagging, collaborative filtering).

The goal of this special issue is to gather high-quality and original contributions that reach beyond conventional ideas and approaches and make substantial steps towards dependable, practically deployable semantic inference theories and algorithms.

Topics of interest include (but are not limited to):

- Theory and algorithms of robust, generic, and scalable semantic inference
- Self-learning and interactive learning for online adaptable semantic inference

- Exploration of applicability scope and theoretical performance limits of semantic inference algorithms
- Modeling of system confidence in its semantic inference performance
- Evaluation of semantic inference dependability using standard dependability criteria
- Matching user/context requirements to dependability criteria (e.g., mobile user, user at home, etc.)
- Modeling synergies between different semantic inference mechanisms (e.g., content analysis, indexing through user interaction, collaborative filtering)
- Synergetic integration of content analysis, user actions (e.g., tagging, interaction with content) and user/device collaboration (e.g., in social/P2P networks)

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