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PHYSICS-BASED PARAMETRIC SYNTHESIS OF INHARMONIC PIANO TONES

Jukka Rauhala

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Helsinki University of Technology
Laboratory of Acoustics and Audio Signal Processing
P.O. Box 3000
FIN-02015 TKK
Tel. +358 9 4511
Fax +358 9 460 224
E-mail lea.soderman@tkk.fi
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Abstract

This dissertation studies methods for developing a parametric piano synthesis model using the physics-based approach. The goal is to develop a model that can be controlled with physically meaningful parameters. Moreover, the model is required to be computationally efficient for real-time implementation. The basis of this work is to use the digital waveguide technique for implementing a piano string model. The excitation signal, simulation of dispersion, the beating effect, and simulation of sympathetic resonances are considered. Novel and improved simulation methods are developed for each of these aspects by applying signal processing techniques and knowledge of the human auditory system. The new simulation methods include a novel excitation model with parametric control and the first closed-form design method for dispersion filter design. In addition, two new beating effect simulation methods suitable for parametric real-time synthesis are created. One of the developed methods can be also used for modifying the partial envelopes in recorded tones. Furthermore, an efficient and improved method for simulation of sympathetic resonances has been suggested. Additionally, a novel analysis method for estimating inharmonicity coefficient values from recorded tones, which is needed for high-quality synthesis, is developed giving good results. Finally, a real-time piano synthesis model without any sampled sounds is implemented using the developed simulation methods in collaboration with the Sibelius Academy. The model can be controlled in real-time using physical parameters, such as the fundamental frequency and the inharmonicity coefficient value. The implementation suggests that the goals set for this thesis work are met. The results can be applied to physics-based piano synthesis. The methods can be used to implement a synthesis model for restricted environments, and they can be used to produce test tones for evaluating properties of the human auditory system and testing signal analysis algorithms.

Keywords acoustic signal processing, digital signal processing, music	
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Tiivistelmä

Tämä väitöskirja käsittelee menetelmiä, joiden avulla voidaan luoda parametrinen pianosynteesimalli käyttäen fysikaaliseen mallinnukseen pohjautuvaa lähestymistapaa. Työn tavoitteena on tuottaa malli, jota voidaan ohjata fysikaalisesti tärkeillä muuttujilla. Lisäksi mallin on oltava tarpeeksi kevyt laskennallisesti, jotta se voidaan toteuttaa reaaliajassa. Työn lähtökohtana on aaltojohtotekniikalla toteutettu pianon kielimalli. Pianomallin eri piirteistä tarkastellaan erityisesti herätettä, dispersiota, huojuntaa sekä sympaattista värähtelyä, joiden simulointiin kehitetään uusia ja paranneltuja menetelmiä hyödyntämällä sekä signaalinkäsittelytekniikoita että tietoa ihmisen kuulojärjestelmän piirteistä. Herätteen tuottamiseen on kehitetty uusi menetelmä, jossa herätesignaalia voidaan kontrolloida parametreillä. Dispersioilmiötä simuloivan suotimen suunnitteluun on luotu uusi menetelmä, jolla suodin voidaan ensimmäistä kertaa suunnitella suljetun muodon kaavalla. Huojunnan simulointiin on vastaavasti kehitetty kaksi menetelmää, joita voidaan molempia käyttää reaaliaikaisissa ja parametrisissä malleissa. Toista menetelmää voidaan käyttää myös äänitettyjen äänten harmonisten vaimenemiskäyrien muokkaamiseen. Sympaattisten värähtelyiden simulointiin on puolestaan keksitty uusi, tehokas menetelmä. Lisäksi työssä on kehitetty uusi analyysimenetelmä dispersiosta aiheutuvan epäharmonisuuden mittaamiseen äänitetyistä signaaleista. Tulokset osoittavat että analyysimenetelmä tuottaa hyviä tuloksia. Lopuksi työssä on toteutettu yhteistyössä Sibelius-Akatemian kanssa reaaliaikainen pianosynteesiohjelma, jossa ei käytetä äänitettyjä ääniä. Synteesimallia voidaan ohjata reaaliajassa fysikaalisilla parametreillä, kuten perustaajuudella ja epäharmonisuuden määrällä. Toteutuksella, jossa käytetään tässä työssä kehitettyjä menetelmiä, osoitetaan että työlle asetetut tavoitteet ovat täyttyneet. Työn tuloksia voidaan hyödyntää fysikaaliseen mallinnukseen pohjautuvassa pianosynteesissä. Lisäksi synteesimallista on mahdollista kehittää kevyempi versio ympäristöihin, joissa käytettävissä oleva muistin määrä sekä prosessoriteho ovat rajalliset. Työssä esiteltyjä menetelmiä voidaan käyttää myös tuottamaan testiääniä ihmisen kuulojärjestelmän piirteiden analysointiin ja signaalianalyysimenetelmien testaamiseen.

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Preface

This work has been carried out at the Laboratory of Acoustics and Audio Signal Processing at Helsinki University of Technology (TKK) during the years 2005-2007.

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My wife Maria has been an important help and source of encouragement in this process and I am deeply grateful because of that. Without her support this achievement would not have been possible. Also, I wish to thank little Jonatan, the sunshine of our lives, who has brought much joy into our lives and supported me in this work in his own way. Finally, I wish to thank my heavenly Father who is the source of all wisdom. May this dissertation bring glory to Him, who deserves it.

Espoo, November 2007

Jukka Rauhala

Contents

List of Publications

This thesis consists of an overview and of the following publications, which are referred to in the text by their Roman numerals.

I J. Rauhala and V. Välimäki, "Tunable dispersion filter design method for piano synthesis," *IEEE Signal Processing Letters*, vol. 13, no. 5, pp. 253–256, 2006.

II J. Rauhala and V. Välimäki, "Dispersion modeling in waveguide piano synthesis using tunable allpass filters," in *Proc. 9th International Conference on Digital Audio Effects*, Montreal, Canada, 2006, pp. 71–76.

III J. Rauhala and V. Välimäki, "Parametric excitation model for waveguide piano synthesis," in *Proc. 2006 IEEE International Conference on Acoustics, Speech, and Signal Processing*, Toulouse, France, 2006, pp. 157–160.

IV J. Rauhala, H.-M. Lehtonen, and V. Välimäki, "Toward next-generation digital keyboard instruments," *IEEE Signal Processing Magazine*, vol. 24, no. 2, pp. 12–20, 2007.

V J. Rauhala, H.-M. Lehtonen, and V. Välimäki, "Fast automatic inharmonicity estimation algorithm," *Journal of the Acoustical Society of America*, vol. 121, no. 5, pp. EL184–EL189, 2007.

VI J. Rauhala, "The beating equalizer and its application to the synthesis and modification of piano tones," in *Proc. 10th International Conference on Digital Audio Effects*, Bordeaux, France, 2007, pp. 181–187.

VII J. Rauhala, M. Laurson, V. Välimäki, V. Norilo, and H.-M. Lehtonen, "Physics-based piano synthesizer," Laboratory of Acoustics and Audio Signal Processing, Helsinki University of Technology, Tech. Rep. 84, 2007, (submitted to *Computer Music Journal* on June 29th, 2007).

Author's contribution

Publication I: "Tunable Dispersion Filter for Piano Synthesis"

A dispersion filter is needed to simulate inharmonicity, which is an important phenomenon in piano tones. Existing approaches used either large allpass filters or a cascade of allpass filters, designed offline with computationally demanding methods. Hence, they did not enable real-time control over the inharmonicity parameter. In this paper, a novel method was presented based on the Thiran allpass fractional delay filter design method. The method provided a closed-form formula for determining the filter parameters. Hence, the method provided a simple way to design dispersion filters and it enabled even real-time modification of the inharmonicity coefficient value. The original idea of using the Thiran filter design technique in the method was suggested by the second author. The parameterization and testing was done by the author.

Publication II: "Dispersion Modeling in Waveguide Piano Synthesis Using Tunable Allpass Filters"

A well-known approach for dispersion simulation uses a cascade of first-order allpass filters. However, the filters have to be designed by trial and error, as there is no closed-form design method for these filters. This paper applies the tunable dispersion filter design method published in Publication I for designing first-order allpass cascades. Moreover, the method was parameterized to design cascades with an arbitrary number of filters. The simulation results showed that the method can be used for designing the desired filters. This technique was invented by the author.

Publication III: "Parametric excitation model for waveguide piano synthesis"

An essential part of the waveguide piano model is the excitation model, which sets the partial amplitude levels, thus contributing significantly to the color of the sound. Moreover, the excitation model must also be velocity-dependent in order to provide dynamics to the sound. Existing models did not provide direct control over the partial amplitudes and partial frequencies, which created a problem when, for example, the inharmonicity of the tone was modified. A new method was proposed to solve these problems by using additive synthesis to create the partial signals. In addition, a velocity-controlled band-stop filter designed with a multi-rate technique provided velocity-dependency. Moreover, higher frequencies were produced with highpass-filtered noise in order to decrease the computational load. The new method was shown to be able to produce desired signals according to the partial amplitude and partial frequency parameters. The new method was invented and tested by the present author.

Publication IV: "Toward next-generation digital keyboard instruments"

This paper presents a review on the acoustics and the synthesis of digital keyboard instruments, namely the clavichord, the harpsichord, and the piano. Moreover, a parametric keyboard instrument model, which can be controlled with the fundamental frequency and the inharmonicity coefficient parameters, is introduced. A novel feature in this model is a new kind of beating method, which uses amplitude modulation to produce the beating effect. The author contributed to the latter part that discusses the parametric model. He also invented the new beating method introduced in this paper.

Publication V: "Fast automatic inharmonicity estimation algorithm"

Automatic estimation of the inharmonicity coefficient value from recorded piano tones is a challenging task. Previous methods are time-consuming due to their large iteration loops. In this paper, a novel method was proposed based on an intuitive approach. The key idea was to examine a partial frequencies curve, which gives a good indication of whether the inharmonicity coefficient estimation is higher or lower than the accurate value. The estimation process includes an iteration loop with an adaptive step size. In addition, the same curve can be used to refine the fundamental frequency estimate, which will further improve the final estimate. The results show that the method is able to match the quality of previous methods while being much faster. The new method was invented by the author.

Publication VI: "The beating equalizer and its application to the synthesis and modification of piano tones"

In the beating method presented in Publication IV, the method consisted of a narrow band-pass filter, which separates the desired partial, and a modulating signal that was multiplied with the separated signal. Finally, the resulting signal was mixed with the original signal. This method was further improved in this paper by producing the beating effect by using a single equalizing filter and by modulating its gain value. The new method uses a second-order equalizing filter, which can be structured in such a way that the peak gain depends on a single filter coefficient in a feedforward path. Hence, the method offers a direct relation between the beating effect parameters and filter coefficients. The results show that the method is able to produce the desired beating effect. Moreover, the method can be used for modifying recorded tones by increasing or canceling the beating effect. The new method was invented by the author.

Publication VII: "Physics-based piano synthesizer"

In this paper, a piano synthesis model incorporating the methods presented in Publications I, III, and IV is proposed. A novel method for producing sympathetic resonances is presented, extending a previously introduced method designed for the acoustic guitar. Moreover, the complete model is implemented by using a real-time software, PWGL. It is shown how the model parameters, including the fundamental frequency and inharmonicity coefficient, can be fine-tuned in real-time. The new method for simulating sympathetic resonances was invented by the author.

List of Abbreviations

DWG	digital waveguide
FD	finite difference
LSEE	least-squares equation-error
LTI	linear and time-invariant
PFD	partial frequencies deviation
STFT	short-time Fourier transform
WDF	wave digital filter

List of Symbols

$A(z)$	transfer function of an allpass filter
a_k	allpass filter coefficient
a_{op}	one-pole filter coefficient
b	multi-ripple filter gain coefficient
B	inharmonicity coefficient value
$\hat{B}$	B estimate
C_{d}, C_1, C_2	tunable dispersion filter constants
D_k	partial frequencies deviation curve
$d_{\mathrm{disp},1}$	delay at the fundamental frequency introduced by the dispersion filter
D	target phase delay of the allpass filter at dc
f_0	nominal fundamental frequency
f_1	fundamental frequency
$f_{\mathrm{b}}, f_{\mathrm{bw}}$	bandwidth of the passband in the equalizing filter
f_c	center frequency of the passband in the equalizing filter
$\hat{f}_k$	partial frequency estimate
f_{s}	sampling frequency
$G_{\mathrm{b}}, g_{\mathrm{b}}$	beating depth of the partial beating method
g_{c}	beating method coefficient
g_{m}	overall equalizing filter gain
G_{min}	minimum gain of the one-pole filter in the excitation method
g_{n}	gain of the notch filter in the excitation method
G_{op}	gain of the one-pole filter in the excitation method
$H_{\mathrm{bp}}(z)$	transfer function of a bandpass filter
$H_{\mathrm{EQ}}(z)$	transfer function of an equalizing filter
H_i	hammer knocking tone simulation block
$H_{\mathrm{MR}}(z)$	transfer function of the multi-ripple filter
$H_{\mathrm{w}}(z)$	transfer function of the windowing function

I_{key}	piano key index
k	partial index
K	notch gain of the equalizing filter
$k_{\mathrm{d}}, k_1, k_2, k_3$	tunable dispersion filter constants
$K_{\max}$	maximum number of partials
l	length of the string
L	length of the windowing function
L_1	length of the delay line when using a dispersion filter
M	number of allpass filters in a cascade
$m_1 - m_4$	first-order tunable dispersion filter constants
N	dispersion filter order
N_{b}	number of feedforward branches in the multi-ripple filter
N_{i}	number of partial beating models
N_{p}	maximum number of simultaneous notes
N_{s}	number of active strings
P_k	target phase delay of the feedback loop at the frequency of partial k
Q	Young's modulus
r_i, l_i, m_i	sympathetic resonance simulation coefficients
r_n	gain of the feedforward path
R_n	length of the feedforward path in samples
$S_{i,k}$	string model block
v	key press velocity
w	Hanning window values
y_{LFO}	LFO signal
δ	adaptive step size for B estimation
μ	adaptive step size for f_1 estimation

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1 Introduction

This thesis discusses sound synthesis of the piano using physics-based methods. Sound synthesis can be categorized under the music technology research field. Moreover, in order to develop sound synthesis methods, sound analysis, which belongs to the group of musical acoustics, is done along with sound synthesis. The term physics-based sound synthesis, or physically inspired sound synthesis, can be used to describe a synthesis method, which uses physical models and signal processing methods for simulating phenomena occurring in the sound [3, 4, 5].

The two most common physical modeling techniques used in sound synthesis are the digital waveguide technique (DWG) [6, 7] and the finite difference method (FD) [8, 9]. The DWG technique can be used to develop simplified and efficient physical models, whereas the FD method produces models that are directly derived from real physical parameters at the cost of increased computational load compared to the DWG.

In this work, a piano synthesis model is developed using the DWG technique. The main goal is to develop a parametric model that can be controlled in real-time by computing model coefficients on the fly. In addition, the model is required to be efficient without compromising the perceived sound quality. Another aspect of this work is to pay special attention to the inharmonicity phenomenon, which is a characteristic of the piano.

1.1 Scope of the research

The primary goal for this thesis is to develop methods that enable the implementation of a parametric, real-time piano synthesizer allowing dynamic online control over the model parameters. Elements that are considered include the excitation model, dispersion filter, and beating simulation. Special attention is paid to the fundamental frequency and inharmonicity coefficient parameters that affect most of the components in the model. Moreover, a method for simulating sympathetic resonances in a piano synthesis model is considered. Finally, a real-time implementation based on the developed methods is created. Another goal in this work is to develop methods that simulate the desired phenomena in an efficient way suitable for real-time implementation. An important aspect that can be used in optimizing the methods is the human auditory system and its properties. In other words, it is better to direct the memory and computational resources to parts that are perceptually important. Moreover, the model can be simplified and stripped down in places that do not affect the perceived sound. In addition to the perceptual information [10, 11, 12, 13, 14, 15], also signal processing techniques are used for improving simulation methods.

1.2 Main results

This thesis work has produced the following results:

- The first closed-form method for designing dispersion filters using first- or second-order allpass filters.
- A new parametric excitation model, which provides a direct control of the simulation parameters, such as the fundamental frequency and the inharmonicity coefficient value.
- A novel inharmonicity estimation algorithm that is efficient and robust.
- Two new beating simulation techniques that offer direct control of the beating parameters. One of these techniques can be used for both synthesis and analysis purposes, as it can be used to modify partial envelopes of arbitrary tones.
- A new method for simulating sympathetic resonances in a piano synthesis model.

- A real-time piano synthesizer developed in co-operation with the Sibelius Academy.

The main area where these results can be applied is physics-based piano synthesis. Due to the highly parametric methods, the model can be scaled down to be used in restricted environments, such as mobile phones and game platforms. Moreover, the real-time parametric control over the fundamental frequency and inharmonicity coefficient parameters offers a piano model, where these parameters can be tuned in a similar fashion as the fundamental frequencies are tuned in a real grand piano. Other application fields for piano synthesis are signal analysis, such as transcription of piano music [16, 17, 18], and analysis of the perception [10, 11, 12, 13, 14], where it can be used for producing realistic test tones. Additionally, the inharmonicity estimation algorithm can be used for signal analysis, namely for estimating fundamental frequencies [19] and inharmonicity coefficient values from recorded string instrument tones.

2 Background

2.1 Acoustics of the piano

The piano is one of the most popular instruments in western music. It is an essential instrument in many musical styles ranging from jazz to classical and pop. The modern piano has evolved from the 18th century predecessor called "gravicembalo col piano et forte", which can be considered as a modified harpsichord [20]. Today, there are two types of pianos available: grand pianos and upright pianos. Grand pianos, where the strings are laid horizontally, are popular in the professional context due to excellent sound quality, whereas upright pianos with vertical strings are suitable for homes as they are smaller and cheaper to purchase. Another advantage that the grand piano has over the upright piano is that it enables fast repetitions on the same note. This work concentrates on the grand piano.

The basic structure of the piano, which is depicted in Figure 2.1, includes the keyboard, the action, the strings, the soundboard, and the frame. Each piano key is connected to a hammer and a damper via a mechanism inside the action. In the rest position, the damper lies on the string and the hammer is ready to strike the string group. When the key is pressed downward, the damper is lifted releasing the string group to vibrate freely, and the hammer strikes the strings with a force depending on the velocity of the press. As the key is released to the rest position, the damper lands on the string group. Each group of strings has one, two, or three strings depending on the key. The lowest eight keys correspond to single strings, whereas the 59 highest keys are attached in groups of three strings. Altogether, a grand piano has 243 strings.

From the synthesis point of view, the piano has two special features compared to other popular instruments with strings. First, there is no direct contact between the user and the

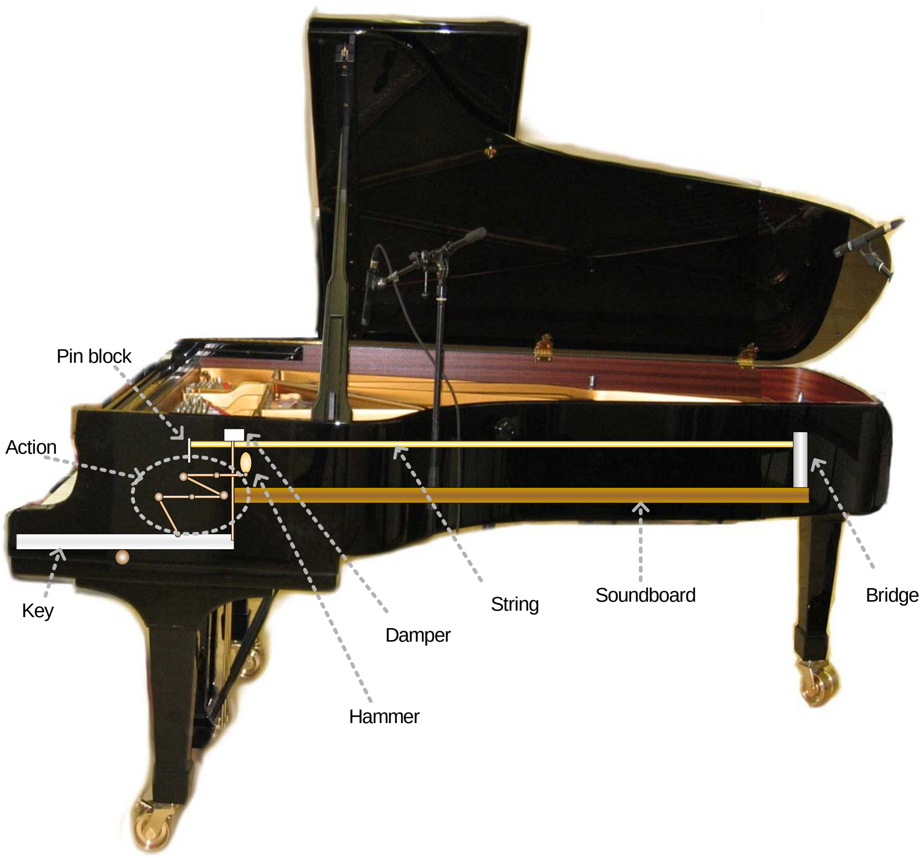

Figure 2.1: *Structure of the piano.*

strings, for example as in the guitar, making the excitation process from the sound synthesis point of view somewhat easier. On the other hand, the piano has a very complex structure with over two hundred strings, which is very challenging for the synthesis. The most significant phenomena from the piano tone point of view are dispersion, two-stage decay, beating, frequency-dependent decaying, and phantom partials. The two-stage decay means that in the beginning the piano tone decays rapidly but after some tenths of a second the decaying slows down. This is explained by the shift in the dominating vibration from vertical to horizontal [21]. Moreover, the phantom partials effect is a phenomenon occurring in the fortissimo piano tones due to longitudinal modes [22, 23, 24]. The general acoustics of the piano is well described, for example in [25, 26, 27, 28], while details on the hammer-string interaction are available in [29, 30, 31, 32, 33, 34, 35, 36, 37, 38], and the acoustical phenomena related to piano strings are presented in [39, 40, 41, 42]. Additionally, the mechanics of the piano soundboard and its effect on the piano acoustics can be found in [43, 44, 45, 46]. Also, characteristics of a piano tone are described in [47, 48, 49].

2.2 Physics-based sound synthesis of the piano

2.2.1 Physics-based methods in sound synthesis

In physics-based sound synthesis [3, 50, 4, 5, 51], the goal is to simulate the sound source instead of just the sound itself. Currently, there are six synthesis methods that are considered physical modeling techniques: finite-difference (FD), mass-spring networks, wave digital filters (WDF), modal synthesis, source-filter modeling, and digital waveguide (DWG) [52, 53]. In practice, there are two main approaches in applying physical modeling techniques. The first approach prefers to use strict physical modeling, while the other concentrates on developing computationally efficient physics-based methods. This is done by using physical modeling as the basis for the synthesis and by applying signal processing methods to improve computational efficiency. This work uses the latter approach.

One of the great advantages of the physics-based approach is its support for parametric control of the model. While the sampling synthesis technique can be used to produce high-quality piano tones, it does not offer the same kind of flexibility as physics-based models. For instance, in order to change the pitch of a string, the sampling technique needs additional signal processing methods to perform the change. On the other hand, the fundamental frequency is one of the main internal parameters in physics-based string models and, hence, it can be easily modified offline. Online control over parameters introduces challenges that are dealt in this work.

FD modeling is based on the numerical solution of partial differential equations [8, 9, 54]. It can be used to develop very accurate physical models. Moreover, it maps physical parameters directly to the equations. The major disadvantage in sound synthesis is its computational inefficiency [55]. Another inefficient physical modeling technique is mass-spring networks, which uses finite masses, springs, and dampers to define a physical system [56]. The WDF method has been also used for sound synthesis. It was originally developed for representing electrical analog circuits in the digital domain with equivalent elements [57], but later it has been applied to other domains, including the acoustical domain enabling sound synthesis [58]. In modal synthesis, the goal is to simulate the modes of vibrations [59, 60]. Recently, modal synthesis principles have been applied to sound synthesis by using the functional transformation method [61, 62]. The source-filter technique that uses time-variant filters is also considered as physical modeling technique [63, 64]. The DWG technique is presented in Section 2.2.2.

The DWG technique and the FD method are the most common physical modeling techniques used for the synthesis of the piano. Chaigne and Askenfelt have developed a piano synthesis model using the FD technique [65, 66]. Another more recent FD piano simulation has been presented by Giordano and Jiang [67]. Moreover, Bank and Sujbert has used the FD method for simulating longitudinal modes [68]. Furthermore, some work has been done using the source-filter-based approach for piano synthesis [69, 70]. Other techniques have been used as well for describing single subsystems of a piano synthesis model. For instance, Van Duyne and Smith have proposed a hammer model using the WDF technique [71].

2.2.2 Piano synthesis using digital waveguides

The DWG technique, which can be considered as an extension of the Karplus-Strong algorithm [72], is a popular physical modeling technique [6, 7]. It is well suited, for example, for modeling string and wind instruments. It has been applied, for example, to synthesize of the acoustic guitar [73, 74, 75], the bass guitar [76], traditional string instruments [77, 78, 79, 80, 81, 82, 83, 84], woodwinds [85, 86, 87], brass instruments [88, 89], the violin [90, 91, 92, 93], the harpsichord [94], and the clavichord [95].

The DWG technique is perhaps the most common physical modeling technique in physics-based piano synthesis. The first DWG piano model was developed by Garnett [96]. Since then, work in DWG piano synthesis has been done, e.g., by Van Duyne and Smith [97, 98, 99, 100, 101] and Aramaki et al. [102]. Moreover, Bensa and his colleagues have conducted research on piano string modeling [103, 104, 105, 106], hammer-string interaction [107, 108, 109], and phantom partials [110]. Also, Bank and his co-authors have worked on the nonlinearities in DWG piano modeling [111, 112, 113], on the simulation of the beating effect [114, 115, 116], on loss-filter design [117], and on the modeling of longitudinal modes [24, 118, 68]. Additionally, excellent reviews on the DWG piano synthesis are available in [119, 120].

The basis of the DWG technique is the discretization of the traveling-wave equation

$$y = f_1(x - ct) + f_2(x + ct), \tag{2.1}$$

where f_1 and f_2 describe two waves traveling in opposite directions, x is the location on the string, c is the speed of sound, t is time, and y is the displacement of the string from its rest position. This is further illustrated in Figure 2.2. Figure 2.3 shows a DWG model simulating an ideal string with rigid terminations [7]. In other words, each traveling wave f_1 and f_2 corresponds to an equal-length delay line. The external force in Figure 2.3 refers to the force that excites the string to vibrate, namely, in this case, the hammer strike.

When the losses and the dispersion occurring in a real string are taken into account, the model includes L dispersion blocks and loss blocks equally distributed along the delay lines, where N_{L} is the total length of the delay lines in samples. However, if the output of the string system is taken at a single point, these blocks can be combined using linear and time-invariant (LTI) principles into a single dispersion block and a loss block. Moreover,

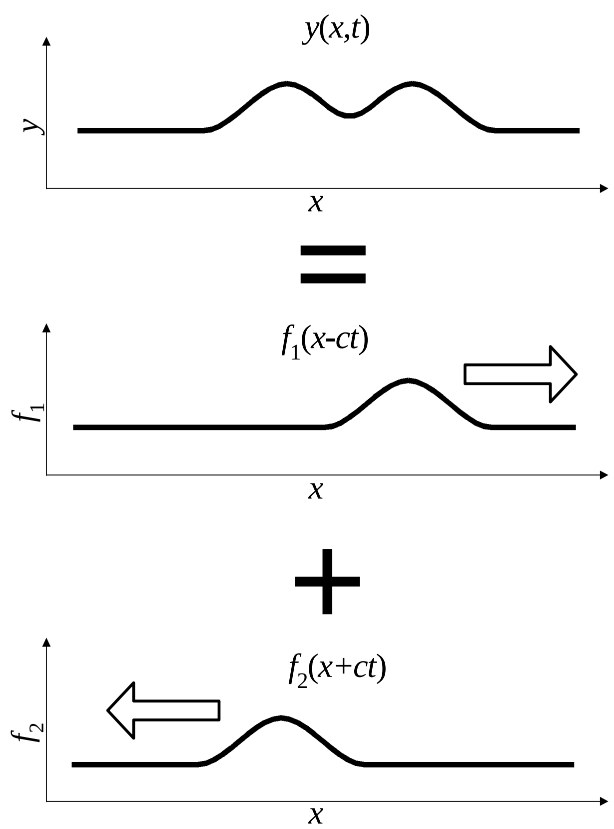

Figure 2.2: *Illustration of the traveling-wave principle. The wave (or the displacement of the string at a certain location) within the string (top) can be described as a sum of two waves traveling in opposite directions (middle and bottom).*

the two delay lines can be combined, and the bridge multipliers can be ignored, as they cancel out each other [121]. As a result, a simplified single-delay line DWG model is obtained, as shown in Figure 2.4.

Commuted waveguide synthesis is a variation of the DWG technique [122, 78]. It applies the LTI principle to DWG models in a way that the order of the serial blocks can be changed without affecting the output of the model. For example, the soundboard block, which can be considered as the last block in the piano model, can be moved between the excitation and the string blocks. In fact, it can be merged with the excitation signal. A typical way to build a commuted string instrument model is to obtain an excitation

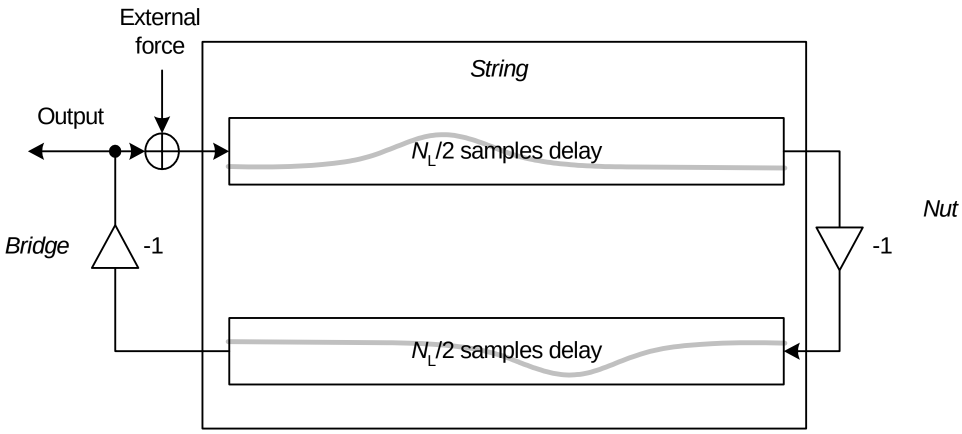

Figure 2.3: *A simple DWG model simulating the two traveling waves in a string with rigid terminations. The losses and the dispersion phenomenon are not taken into account in this model example.*

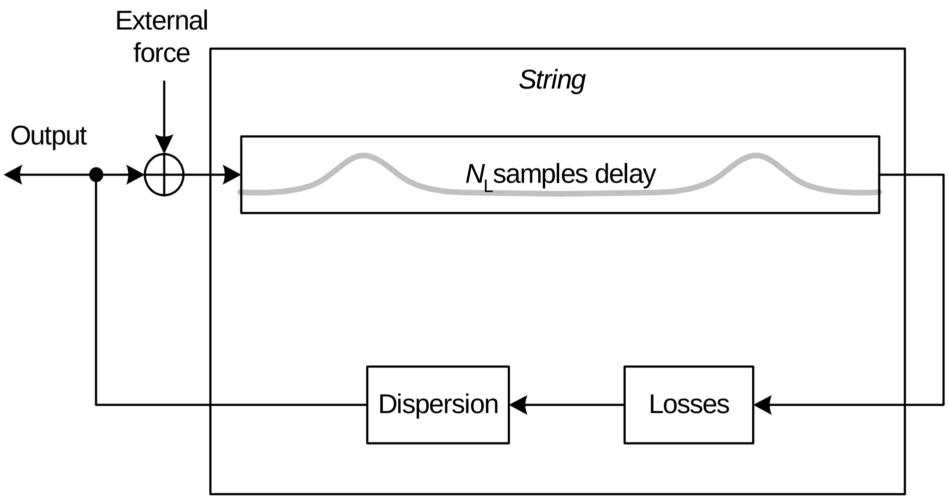

Figure 2.4: *A simple DWG model simulating the two traveling waves with a single delay line. The losses and the dispersion effect are lumped into single points in the feedback loop.*

signal from recorded tones via inverse filtering [51, 73], or by using similar methods [123]. In inverse filtering, the string model is considered to be an IIR filter, which is required to produce the recorded tone when the correct excitation signal is filtered with

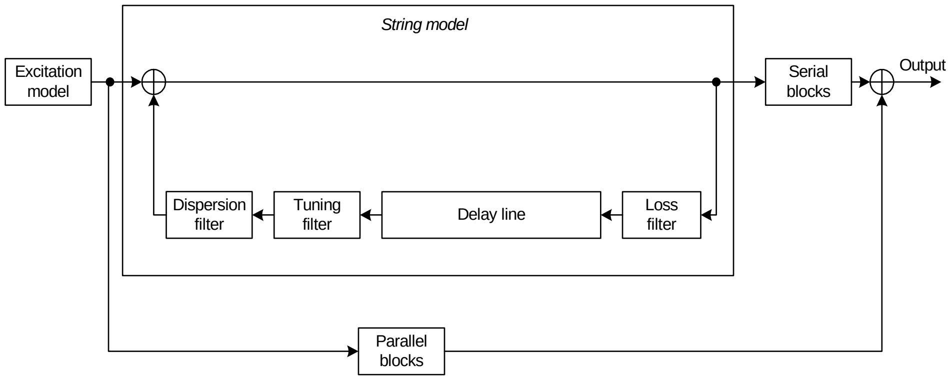

Figure 2.5: *A block diagram of the piano string model using the DWG technique.*

it. Hence, the desired excitation signal can be obtained by filtering the recorded tone with an inverse FIR filter based on the IIR filter, and there is no need for the soundboard model as the effect of the soundboard is included in the excitation signal. However, this approach does not fit well into the parametric piano model, since the excitation signal is not easily controlled using parameters. For instance, if the inharmonicity of the string were modified, a modification in the excitation signal would be required. In the commuted approach using inverse filtering, it would mean that the excitation signal should be redone with inverse filtering. Hence, the commuted approach is not used in this work as such.

Figure 2.5 shows a block diagram of a piano string simulation using the DWG technique. This model uses a signal-based excitation method, as it allows to control individual partial amplitudes. Another approach for simulation would be to use physical hammer models that require bidirectional connection with the string model due to hammer-string interaction, but it does not provide means for controlling individual partial amplitudes. The core string model includes a loss filter, a delay line, a tuning filter, and a dispersion filter. The loss filter is usually a lowpass FIR- or IIR-filter that sets the decay rates of the string output tone [100, 124]. The tuning filter, which is commonly a fractional delay filter [125], is needed in order to tune the string model accurately, since the delay line produces a delay that is an integral multiple of the sampling interval. Finally, the dispersion filter simulates the dispersion phenomena meaning that the phase delay response of the feedback loop must be frequency-dependent [98, 126]. The dispersion filter is usually an allpass filter. In addition to these blocks, the string simulation has an excitation model, parallel blocks, and serial blocks. The role of the excitation model is to simulate the hammer strike exciting the string to vibrate. The parallel and serial blocks denote simulations of the remaining phenomena, including beating [116, 115], phantom partials [110], the sustain pedal [127, 128, 129], and the soundboard [116]. Most of these simulations are implemented as serial blocks, whereas the beating effect can be simulated as a parallel block or a serial block.

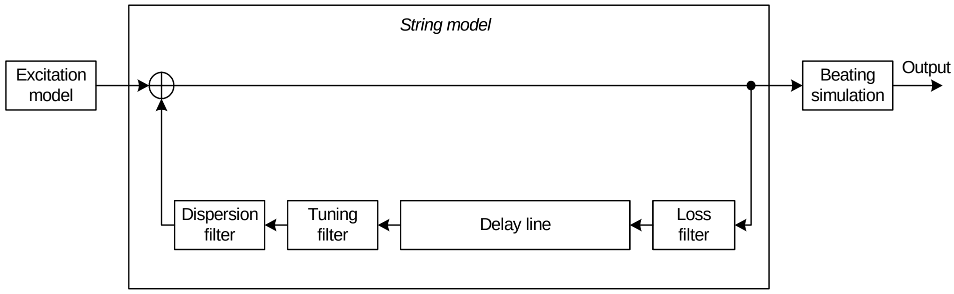

Figure 3.1: *Block diagram of the basic DWG piano string model considered in this work.*

3 Novel piano synthesis model

3.1 Basic piano string model

The basic DWG piano string model considered in this work is shown in Figure 3.1. The main goal of this work is to develop methods that enable real-time control over the fundamental frequency f_0 and the inharmonicity coefficient B parameters in the synthesis model. This type of synthesis model allows the tuning of the fundamental frequencies in a similar way as in a real grand piano. It also enables the fine-tuning of the inharmonicity value. A problem in modifying the inharmonicity value of the piano string is that it affects the perceived pitch as well [14, 130]. Hence, it is important to provide a model that can be tuned by ear in real-time. Table 3.1 presents the effects on each component considered in this work due to the parameterization. The tuning filter used in the model is a conventional allpass fractional delay filter [6, 125, 131], and the loss filter incorporated in the model is a multi-ripple filter [124, 132].

Component	Requirement
Excitation model	Frequencies of the produced partial amplitudes must depend on f_0 and B
Dispersion filter	The phase delay response must depend on f_0 and B
Loss filter	The magnitude response at the partial frequencies, which depend on the f_0 and B, must be as desired
Beating model	Frequencies of the target partials must depend on f_0 and B. Also, slight inaccuracies in partial frequencies due to the dispersion filter must be tolerated

Table 3.1: *Piano model components and the requirements for each component to enable the real-time control of parameters f_0 and B.*

3.2 Simulation of the string excitation

The purpose of the excitation signal is to transfer energy to the string model causing it to vibrate. In other words, the excitation signal simulates the hammer strike. Moreover, the knocking sound occurring when the hammer hits the string and simultaneously excites the soundboard should be simulated as well.

The requirement of parametric control over f_0 and B rules out the use of an inverse-filtered excitation signal, as it cannot be easily modified according to the parameter values especially if the B value is changed. Physical hammer models [133, 134] provide control over f_0 and B, but they do not allow control over individual partial amplitudes. Hence, a novel signal-based excitation method is proposed in [PIII]. The main idea in the new method is to use additive synthesis in producing energy at the partial frequencies. This method can be easily controlled with f_0 and B parameters, as there is direct control over the partial amplitudes.

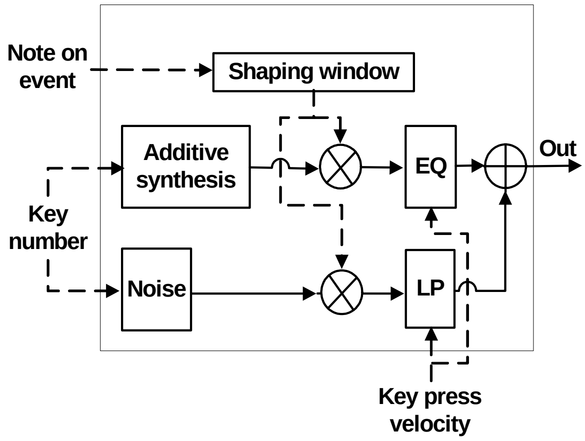

Figure 3.2: *Block diagram of the excitation model.*

The block diagram of the proposed method is shown in Figure 3.2. The method consists of five main blocks: an additive synthesis generator, a noise generator, an equalizing filter, a lowpass filter, and a shaping window. The additive synthesis block produces a sinusoidal signal for each partial according to the desired partial amplitude and frequency. In order to reduce the computational load, the partials with large indices are produced with bandlimited white noise. The model also includes a velocity-controlled equalizing filter for the additive signal in order to produce dynamics in the sound. Similarly, the noise signal is filtered with a velocity-controlled one-pole filter. Additionally, the edges of both source signals are windowed with a Hanning window in order to prevent artifacts in the produced tone caused by sharp edges. The details on the excitation model can be found in [PIII].

In this work, the excitation block parameters are determined semi-automatically. The filter parameters of the velocity-dependent equalizing filter and lowpass filter are obtained by comparing the spectra of recorded pianissimo and mezzoforte piano tones to the spectrum of forte piano tones and by fitting the parameters to produce similar effects on the spectrum depending on the key velocity. The partial amplitudes are determined automatically by analyzing recorded forte piano tones with the short-time Fourier transform

(STFT). Then, the maximum value of each partial envelope is set to be the partial amplitude. Finally, the extracted partial amplitudes are evaluated and, if needed, modified manually.

Piano sound can be separated into a harmonic component (caused by the excited string) and a broadband component (the knocking sound) [128, 47, 135]. The knocking sound is not audible in the bass range, but in the treble range it plays a significant part in the perceived piano sound [136]. Hence, a knocking sound simulating the sound caused by the hammer striking the strings is required in addition to the excitation signal. The knocking sound can be included in the excitation signal, or it can be summed into the signal produced by the string model. The latter option is used in this work, since it does not require any compensation in the partial amplitudes due to the spectral components in the knocking sound. The sound can be produced by using the modal synthesis-based approach presented in [PVII]. An alternative approach is to use processed recorded tones such as an inverse-filtered piano tone.

3.3 Simulation of dispersion

3.3.1 Analysis of the dispersion phenomenon

Piano strings are known to be dispersive due to the stiffness of the string material. Dispersion means that these properties resist the free flexible movement of the string. It is suggested that the soundboard impedance contributes to the inharmonicity as well [137]. As a result, high frequency components in the tone travel faster than low frequency components making the produced tone inharmonic. This is illustrated in Figure 3.3, which is produced by using a short time window [138]. Although inharmonicity is strongest in the treble range of the piano, it is perceptually most important in the bass range, where it adds warmth to the sound [139, 140].

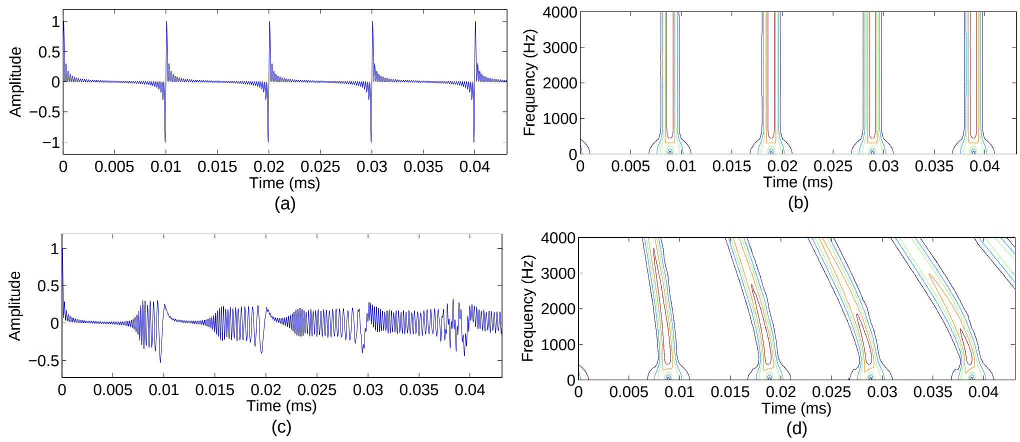

Figure 3.3: *An illustration of how the dispersion phenomenon affects the time-frequency properties of the harmonics. (a) The waveform and (b) the spectrogram of a harmonic tone ($f_0 = 100Hz$, $B = 0$). (c) The waveform and (d) the spectrogram of an inharmonic tone ($f_0 = 100Hz$, $B = 0.0001$).*

Fletcher et al. [139] proposed that the partial frequencies of an inharmonic tone can be computed as

$$f_k = kf_0\sqrt{1 + Bk^2}, \tag{3.1}$$

where k is the partial number, f_0 is the nominal fundamental frequency of the ideal string (non-dispersive), and B is the inharmonicity coefficient defined as

$$B = \frac{\pi^3 Q d^4}{64 l^2 T}, \tag{3.2}$$

where Q is Young's modulus, d is the diameter of the string, l is the length of the string, and T is the tension of the string.

An important part of the analysis of the dispersion phenomenon in the piano is the estimation of inharmonicity coefficient values from recorded piano tones. Trivial methods, such as fitting a curve using Equation 3.1 to the determined partial frequencies, can pro-

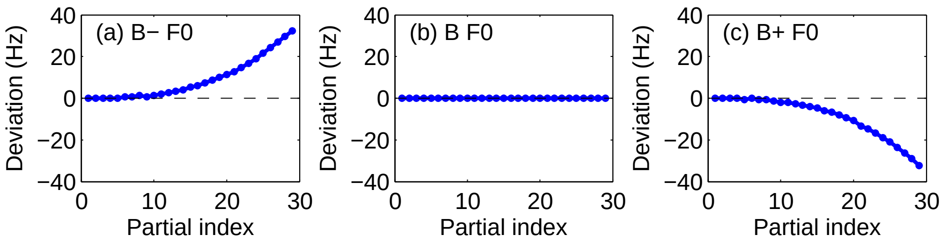

Figure 3.4: *Examples of how the PFD curve behaves in three situations: too low a B estimate value (left), an accurate B estimate value (middle), and too high a B estimate value. The f_0 estimate is accurate in all cases.*

duce in many cases suggestive results. However, these methods are prone to estimation errors that occur when outliers are interpreted as partials, which happens easily as piano tones have a rich spectrum. Previous advanced methods, such as techniques proposed by Galembo and Askenfelt [141, 142], produce fairly good results in an inefficient way. A new solution to this problem is the partial frequencies deviation (PFD) method that is proposed in [PV]. The main idea in this method is to examine a partial frequencies deviation curve, which can be used to determine the quality of a B estimate. The PFD curve can be obtained by calculating the difference between the expected partial frequencies, which depend on the f_0 and B estimates, and the frequencies of dominant spectral peaks found in the spectrum close to the expected frequencies. Assuming that the f_0 estimate is accurate, an increasing PFD curve indicates too low an estimate value, while a decreasing PFD curve suggests too high an estimate value, as seen in Figure 3.4. This property can be used to improve the B estimate through an iteration loop with an adaptive step size, as seen in Figure 3.5. Figure 3.6 shows the block diagram of the PFD method.

In addition to the inharmonicity value, the PFD method also provides information on the quality of the f_0 estimate. The trend of the PFD curve is determined by calculating the signs of its derivative values at each partial index, and by computing the sum of all derivative signs. This leads, most likely, to the three possible situations shown in Figure 3.7: a flat, convex, or concave PFD curve. A flat curve indicates an accurate f_0 estimate,

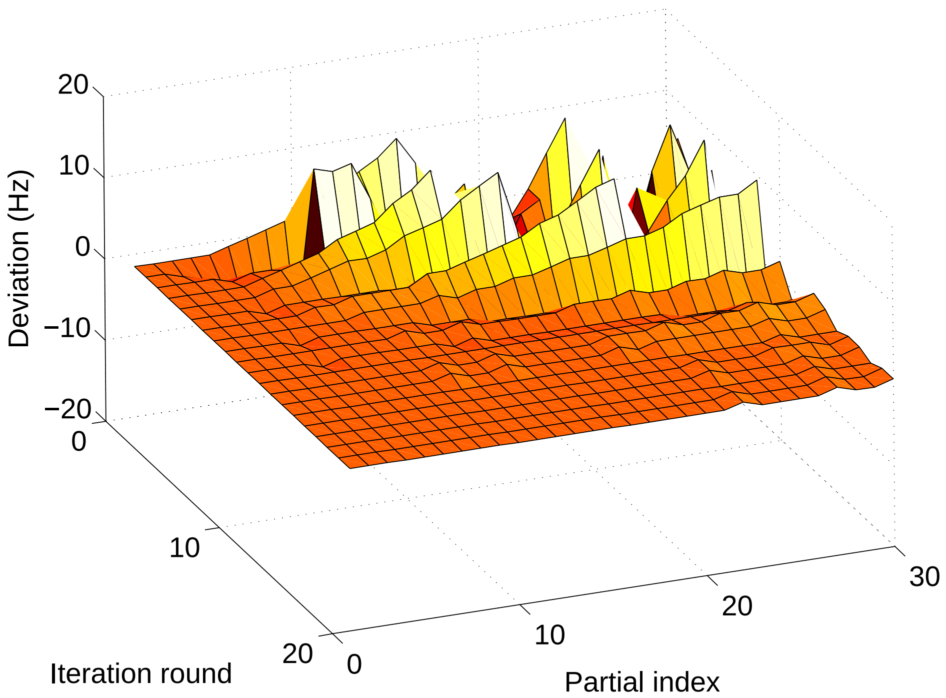

Figure 3.5: *An example of how the PFD method progresses in the iteration process using a synthetic input signal (f_0=38.9 Hz, B=0.0003). This figure shows the PFD curve (deviation as a function of partial index) in iteration rounds 1–20. The deviation is large in the beginning, but is reduced significantly after several iteration rounds. The last deviation curves are very smooth indicating that the modified B estimate value (B estimate at iteration round 20 is 0.000299) is close to the target value.*

whereas a convex curve suggests too low an estimate value and a concave curve hints at too high an estimate value. The inharmonicity estimation process can be improved by refining the f_0 estimate in a similar iteration loop as with the B estimate after running the PFD iteration once, and re-running the PFD iteration with the improved f_0 estimation. Additionally, the PFD method can be used for f_0 estimation of inharmonic piano tones [19].

The results from the test cases presented in [PV] show that the PFD method produces good estimates without heavy computation. Moreover, it is very robust, because single outliers in the PFD curve do not affect the sum of the derivative signs (this is because an outlier

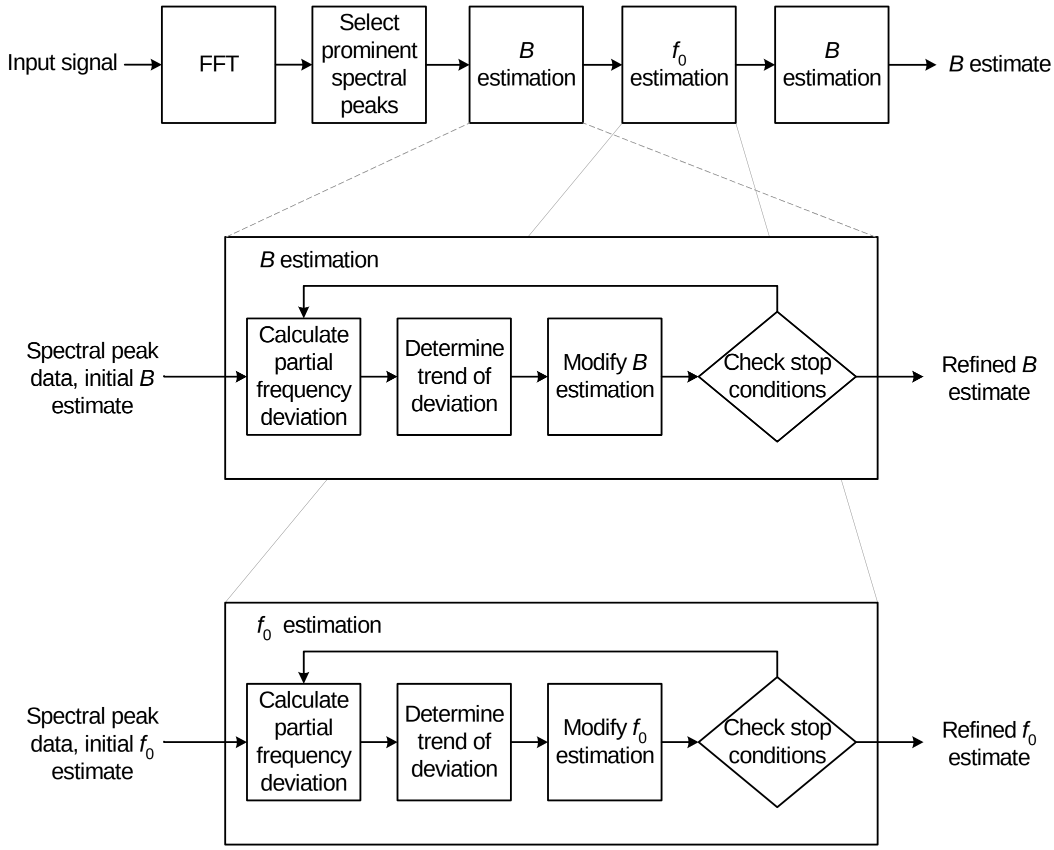

Figure 3.6: *Block diagram of the PFD method.*

produces opposite derivative signs, which corresponds to zero in the sum). In addition, the final PFD curve provides a good error indicator, as an almost flat curve suggests successful estimation and a dispersed curve indicates inaccurate estimation. Figure 3.8 shows an example of inharmonicity values estimated from recorded piano tones using the PFD method.

3.3.2 Dispersion filter design

The dispersion filter is an essential part of the DWG piano string model. As mentioned previously, it is usually an allpass filter with a nonlinear phase delay response that sim-

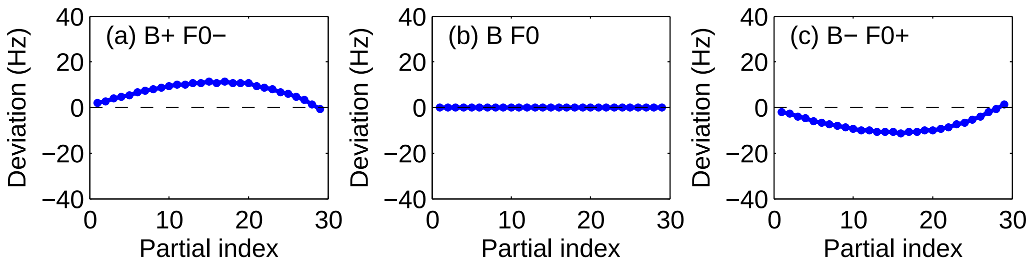

Figure 3.7: *Examples of how the PFD curve behaves in three situations: too high a B estimate and too low an f_0 estimate (left), accurate f_0 and B estimates (middle), and too low a B estimate and too high an f_0 estimate (right).*

ulates the desired frequency-dependent phase delay characteristic of the feedback loop. The desired phase delay response of the feedback loop in samples can be calculated as

$$P_k = \frac{f_\mathrm{s}}{f_0\sqrt{1+Bk^2}}, \tag{3.3}$$

where f_s is the sampling frequency and f_0 is the nominal fundamental frequency. Figure 3.9 shows an example of the phase delay response curve.

Previously, dispersion filters have been designed with an iterative search method. This design approach cannot be used in this work, as it is not possible to control them in real time using parameters f_0 and B. A solution to this problem is the tunable dispersion filter design method proposed in [PI]. This method offers a closed-form formula for determining filter coefficients using f_0 and B parameters as input values. Hence, it offers real-time control over these parameters.

The main idea in the tunable design method is to use the Thiran allpass fractional delay filter design method [143, 144, 125, 131] as a basis, because there is relation between the f_0 and B parameters and the parameter D, which is the delay introduced by the fractional delay filter at dc. By investigating the behavior of the suitable D value when f_0 and B are

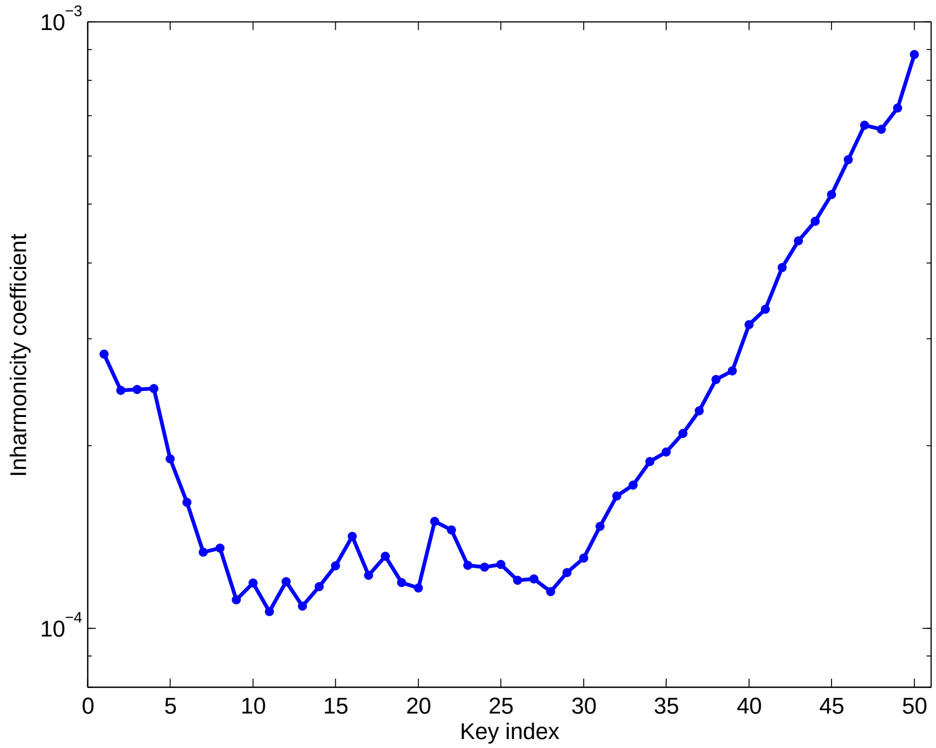

Figure 3.8: *An example of estimated inharmonicity values for key indices 1–50 using the PFD method. Manually estimated f_0 values were used in the estimation. Steinway grand piano samples were obtained from University of Iowa Electronic Music Studios (http://theremin.music.uiowa.edu).*

varied, a closed-form approximating formula for determining D as a function of f_0 and B can be defined as

$$D(I_{\mathrm{key}}, B) = e^{(C_{\mathrm{d}}(B) - I_{\mathrm{key}} k_{\mathrm{d}}(B))}, \tag{3.4}$$

where

$$I_{\mathrm{key}}(f_0) = \log_{\sqrt[12]{2}} \frac{f_0 \sqrt[12]{2}}{27.5}, \tag{3.5}$$

$$k_{\mathrm{d}}(B) = e^{(k_1 (\ln B)^2 + k_2 \ln B + k_3)}, \tag{3.6}$$

$$C_{\mathrm{d}}(B) = e^{(C_1 \ln B + C_2)}, \tag{3.7}$$

and k_1, k_2, k_3, C_1, and C_2 are parameterization constants. In [PI], the dispersion filter was

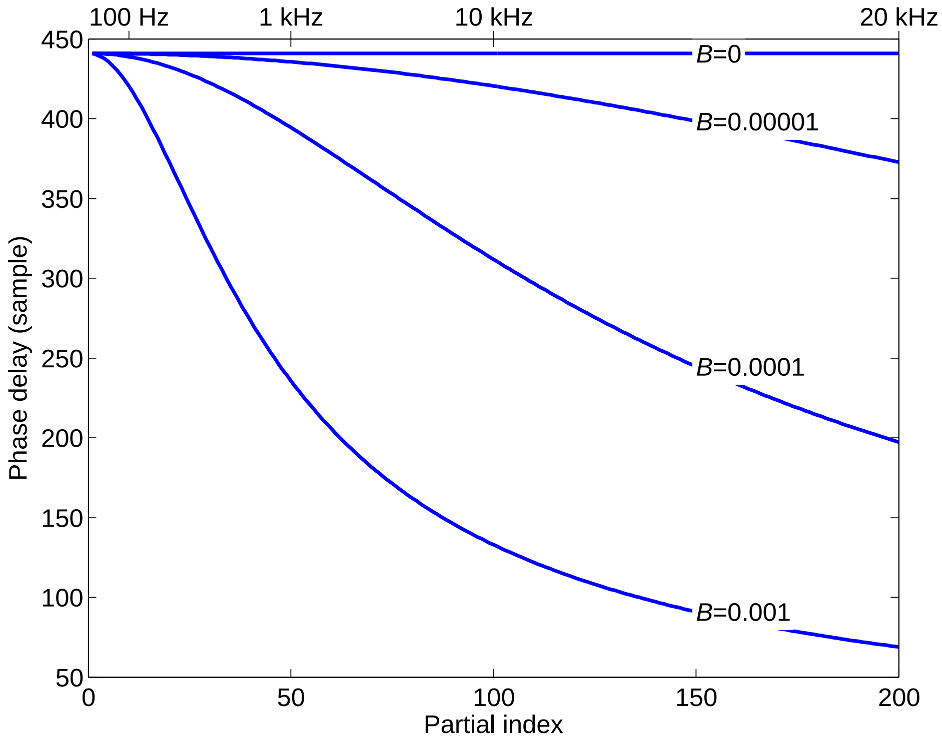

Figure 3.9: *An example of the phase delay response curve of the feedback loop ($f_0 = 100$ Hz) corresponding to B values 0, 0.00001, 0.0001, and 0.001.*

designed to include four second-order filters in cascade for key indices 1–44, and a single second order filter for the rest of the keys. The details of the design method can be found in [PI], and the determined filter parameters are presented in Table I of the paper.

Table 3.2 shows an example of the duration of the filter design with the tunable dispersion filter design method compared to a design method based on the least-squares equation-error (LSEE) method [126]. LSEE is an iterative method for designing allpass filters according to phase delay specification [145] and hence it can be applied to dispersion filter design [126]. Usually, the LSEE algorithm is executed with multiple delay line lengths, as the resulting phase delay response depends on the delay produced at dc. In this example, the delay produced by the dispersion filter at dc is varied from 0 samples to $0.25L$ samples, where L is the total delay of the feedback loop. Moreover, the LSEE algorithm is defined to do 10 iterations at most. After each round, the phase delay response

Method	Design duration (ms)		
	key C_1	key C_2	key C_3
TDF	0.7	0.2	0.2
Iterative LSEE	13985.5	6991.5	3358.9
Single LSEE	29.5	23.0	20.6

Table 3.2: *Design duration times for the tunable dispersion filter design method (denoted as TDF), the complete LSEE-based method (denoted as iterative LSEE), and the simplified LSEE-based method with predefined filter delay at dc and without stability check (denoted as single LSEE) for three keys:* C_1 *(*f_0*=32.7 Hz,* B*=0.00026),* C_2 *(*f_0*=65.4 Hz,* B*=0.00015), and* C_3 *(*f_0*=130.8 Hz,* B*=0.00012).*

needs to be evaluated in order to determine the best value for the delay at dc. In addition, the stability of the filter at each round must be tested, as the LSEE algorithm often produces an unstable allpass filter. In this example, a cascade of two fourth-order allpass filters is designed with the LSEE-based method, as it is computationally comparable to the tunable dispersion filter and it produces better results than four second-order filters.

The results shown in Table 3.2 suggest that the tunable dispersion filter design method outperforms the LSEE-based approach, as the tunable method needs less than a millisecond for the whole process, whereas the latter method requires several seconds. Table 3.2 also uses the case when the optimal delay at dc is known and the LSEE is performed once without checking the stability, which is not a realistic case, but it demonstrates the efficiency of the core algorithm. The results indicate that the LSEE-based approach needs 20-30 ms to produce the filter parameters in the simplified case. Moreover, the results suggest that a significant amount of computation is needed to evaluate the quality of the dispersion filter and to check the stability of the filter at each round. A great advantage with the tunable dispersion filter method is that stability is guaranteed, as the Thiran filter design method always produces stable filters when the produced delay at dc is more than N-1 samples [143], which is the case with this method. Figure 3.10 shows a comparison of the phase delay curves produced with the tunable dispersion filter design method

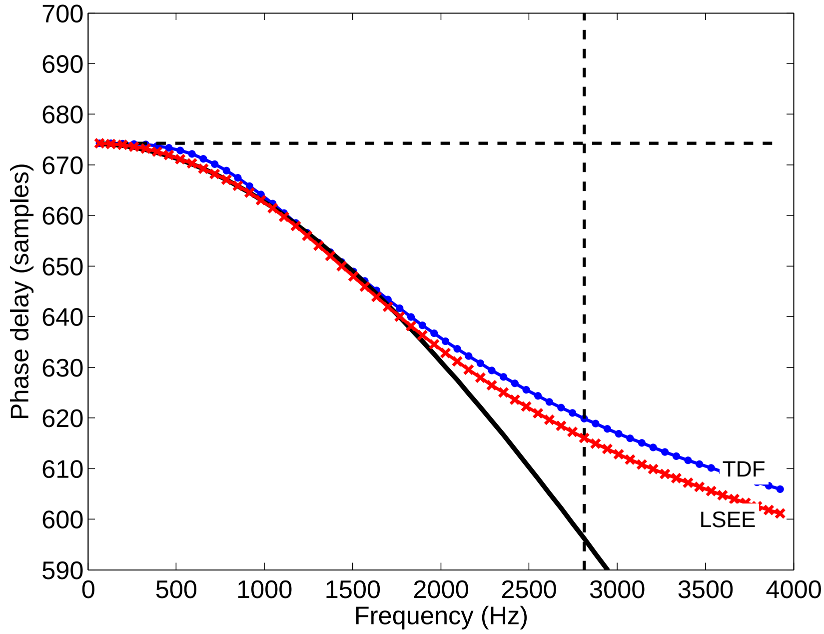

Figure 3.10: *An example of the phase delay response of the second-order tunable dispersion filter with four filters in cascade (solid line with dots) and the phase delay response of a dispersion filter with two fourth-order filters in cascade designed with the LSEE-based method (solid line with crosses) compared to the desired phase delay response (solid line, $f_0 = 65.4$ Hz, $B = 0.00015$). The dashed vertical line is the maximum bandwidth of where the effect is perceived [1] and the dashed horizontal line denotes a harmonic tone. This example corresponds to key C_2 in Table 3.2.*

and the LSEE-based method suggesting that the LSEE-based method produces a slightly better response.

The tunable dispersion filter design method is applied for first-order filters in [PII]. The idea of using a cascade of first-order allpass filters for dispersion simulation was originally proposed by Van Duyne and Smith [98]. However, no closed-form design methods have been introduced. The tunable dispersion filter design method provides a solution to this problem [PII].

In addition to determining new parameters for a first-order filter cascade, the extension proposed in [PII] presents a way to parameterize the number of filters in cascade. This is done by modifying Eq. 3.7 to

$$C_d(B, M) = e^{((m_1 \ln M + m_2) \ln B + m_3 \ln M + m_4)} \quad (3.8)$$

where m_1, m_2, m_3, and m_4 are the polynomial coefficients defined in Table 2 of [PII], and M is the number of filters in cascade. Also, Table 2 in [PII] gives the parameters k_1, k_2, and k_3 for first-order filters. Figure 3.11 shows a comparison of the phase delay responses of the second-order filter and the first-order filter having equal computational cost. Figure 3.12 shows an example of the first-order filter's phase delay response with varying filter cascade size. Details on the design method are found in [PII].

3.4 Simulation of beating

Beating is a phenomenon, which means that certain partial envelopes include modulation. An example of this is seen in Figure 3.13, where significant beating is observed in partials 7, 8, 9, 10, 11, 12, 14, and 15. The main reason for beating is coupling of the string with adjacent strings. Additionally, even single strings can incorporate beating due to a directionally-dependent bridge admittance [146, 21].

A simple but inefficient approach for producing the beating effect is to use two detuned string models [7, 6]. A resonator-based approach [115] is a more efficient method for beating simulation in the case of a few beating partials. However, knowledge of the exact frequency of the partial with the beating effect is required. Otherwise, the beating effect can behave unexpectedly, since in the frequency modulation the modulation frequency depends on the distance of the two spectral component frequencies. On the other hand, the use of a dispersion filter introduces a slight bias to the partial frequencies. Moreover, in the case where the B parameter can be controlled in real time, it is laborious to measure the accurate phase response of the dispersion filter. Hence, frequency modulation-based approaches, such as the use of resonators, is not suitable for this work.

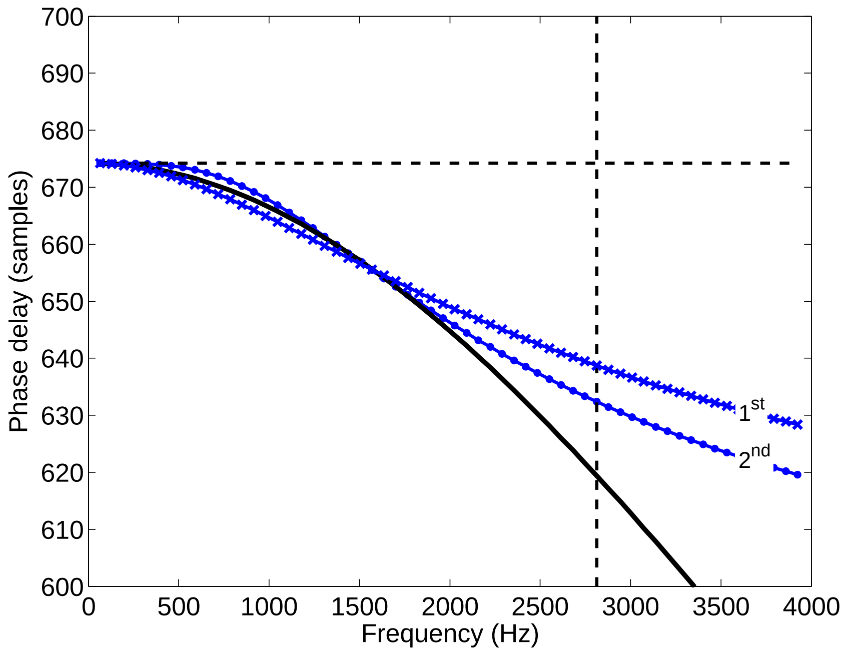

Figure 3.11: *An example of the phase delay response of the second-order tunable dispersion filter with four filters in cascade (denoted as the solid line with dots) and the phase delay response of a corresponding first-order tunable dispersion filter with eight filters in cascade (denoted as the solid line with crosses) compared to the desired phase delay response (denoted as the solid line, $f_0 = 65.4$ Hz, $B = 0.0001$). The dashed vertical line is the maximum bandwidth of where the effect is perceived [1] and the dashed horizontal line denotes a harmonic tone.*

In this work, two alternative amplitude modulation-based beating methods are presented. The basic idea in the method proposed in [PIV] is to produce the beating effect by separating the partial component from the tone with a bandpass filter. Then, the separated signal is modulated with a modulation signal and summed back to the original tone. A block diagram of this method is presented in Figure 3.14.

The other method, introduced in [PVI], produces the beating effect by modulating the gain coefficient of an equalizing filter. The transfer function of the equalizing filter presented by Regalia and Mitra [2] can be written as

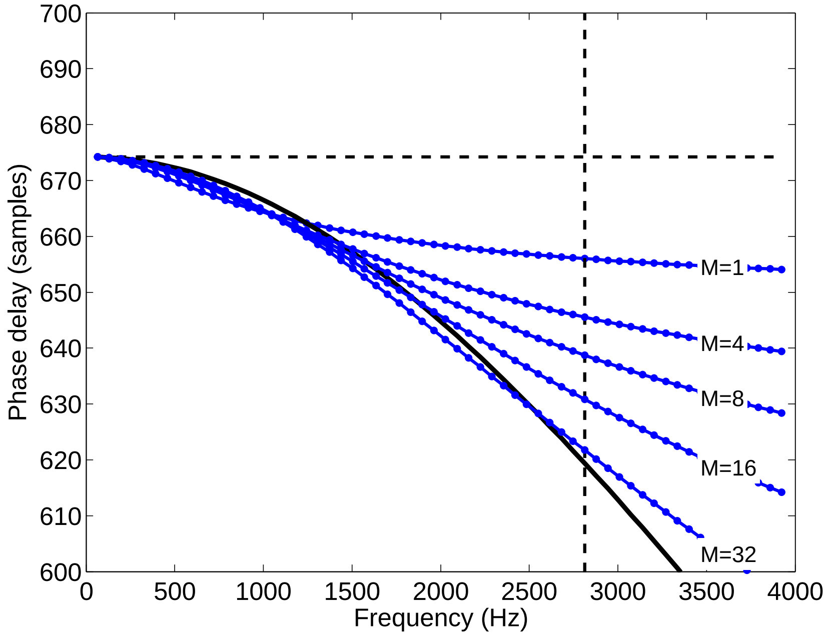

Figure 3.12: *An example of the phase delay response compared to desired phase delay response (denoted as the solid line) of the first-order tunable dispersion filter with varying number of filters in cascade ($f_0 = 65.4$ Hz, $B = 0.0001$). The dashed vertical line is the maximum bandwidth of where the effect is perceived [1] and the dashed horizontal line denotes a harmonic tone.*

$$H(z) = \frac{1}{2}(1+K) + \frac{1}{2}(1-K)A(z), \tag{3.9}$$

where

$$A(z) = \frac{a - \cos(\frac{2\pi f_c}{f_s})(1+a)z^{-1} + z^{-2}}{1 - \cos(\frac{2\pi f_c}{f_s})(1+a)z^{-1} + az^{-2}}, \tag{3.10}$$

$$a = \frac{1 - \tan(\frac{\pi f_{bw}}{f_s})}{1 + \tan(\frac{\pi f_{bw}}{f_s})}, \tag{3.11}$$

f_c is the center frequency of the peak, f_{bw} is the peak bandwidth, f_s is the sampling frequency, and K is the peak gain. In the beating equalizer, f_{bw} is determined as $0.2f_0$, where f_0 is the fundamental frequency. Figure 3.15 shows the block diagram of the proposed beating equalizer. As shown in the figure, the gain of the filter peak can be controlled with the parameter K, which is located in the feedforward path. Hence, the

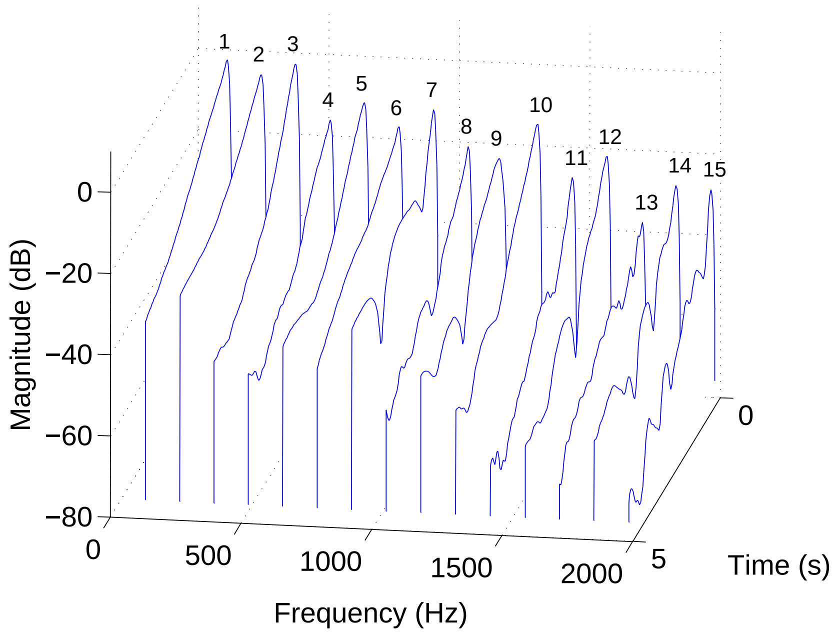

Figure 3.13: *An example of the partial envelopes obtained from a recorded tone (key* C_3*,* f_0*=130.2 Hz). Partial indices are shown above the envelopes.*

filter gain can be easily modulated. A single beating equalizer can produce a beating effect for a single partial. In order to produce a beating effect for multiple partials, multiple beating equalizers can be used in series.

Both of these amplitude modulation-based methods can be used in the parametric piano model, because the dependency of the beating effect behavior on the accuracy of the partial frequency is negligible. Since both methods can use the same modulation signal for producing the beating, the methods are capable of producing partial envelopes similar to each other. The main differences between these methods are that the beating equalizer method offers a simpler structure and a more accurate control over the beating depth, whereas the partial separation method described in [PIV] can use arbitrary modulation signals without causing artifacts due to the fact that the modulated signal is bandlimited.

The beating equalizer can be also used for modifying existing tones. An example pre-

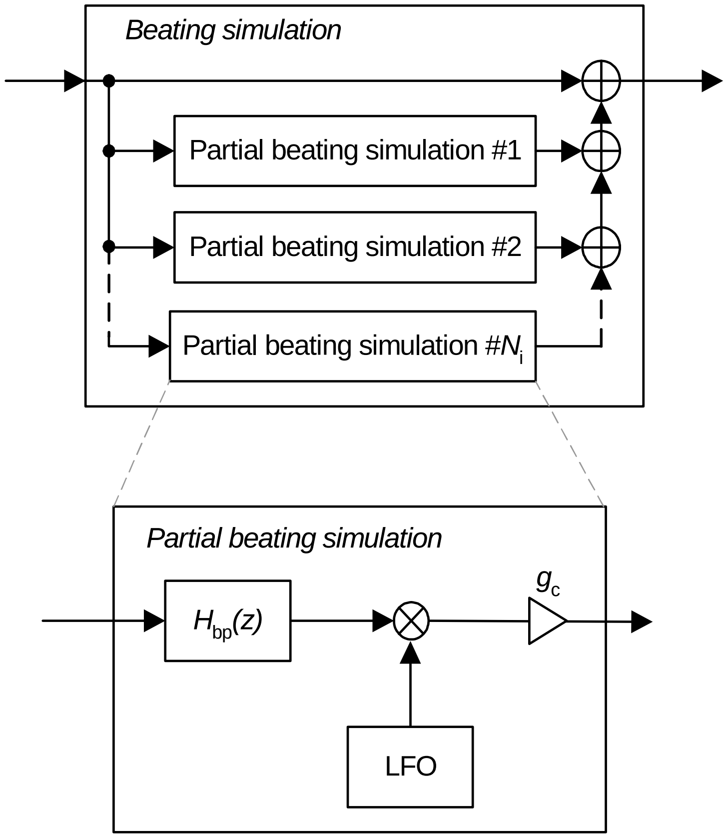

Figure 3.14: *Block diagram of the beating model proposed in [PIV].*

sented in [PVI] shows how the beating equalizer can be used to decrease the beating effect significantly in a recorded piano tone. This is done by analyzing the partial envelopes, and by modulating the original signal with beating equalizers using modulation signals that cancel the beating effect. Moreover, [PVI] gives an example on how the beating effect can be increased in a recorded piano tone.

3.5 Simulation of sympathetic resonances

Until this point, only a single piano string has been considered. When a piano synthesis model with multiple strings is examined, some additional issues will arise. Sympathetic

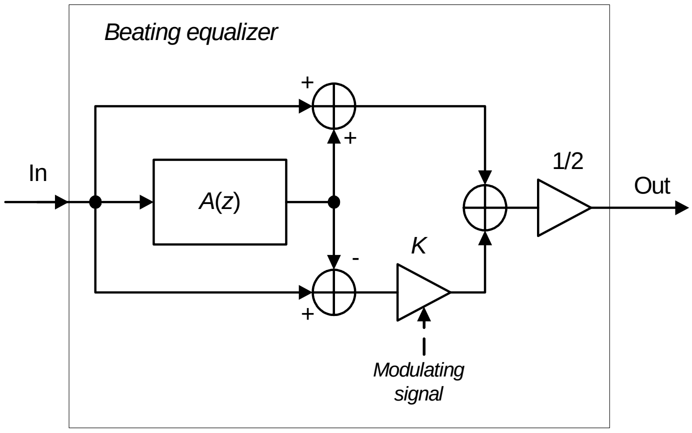

Figure 3.15: *Block diagram of the beating equalizer, which uses the equalizing filter structure proposed by Regalia and Mitra [2].*

resonances is a phenomenon, where vibrational energy is transferred between the undamped strings via the bridge and free air. While it might not be easy to perceive the phenomenon in normal playing, it becomes evident in certain special cases. For instance, if the keys of a certain chord are pressed down slowly without causing audible sound and a forte note is played, it will excite the strings in the chord to produce an audible sound.

Sympathetic resonances is one of the phenomena that cannot be produced with the sampling synthesis technique as such. On the other hand, physics-based sound synthesis offers ways to simulate it. Borin et al. have proposed a method where all active strings are connected to the soundboard impedance filter [129], and the filtered signal is fed back to the strings. A new efficient method for its simulation is proposed in [PVII]. The method extends the idea introduced by Karjalainen et al. for acoustic guitar synthesis [121]. The main idea in the new method is to have two slightly detuned string models, primary and secondary, for each key and to route the sympathetic resonance signal to opposite directions in the primary and secondary strings. Moreover, the primary and secondary string models can be connected by feeding the signal from primary string models to secondary string models. The method includes single delays between the serial string models. In

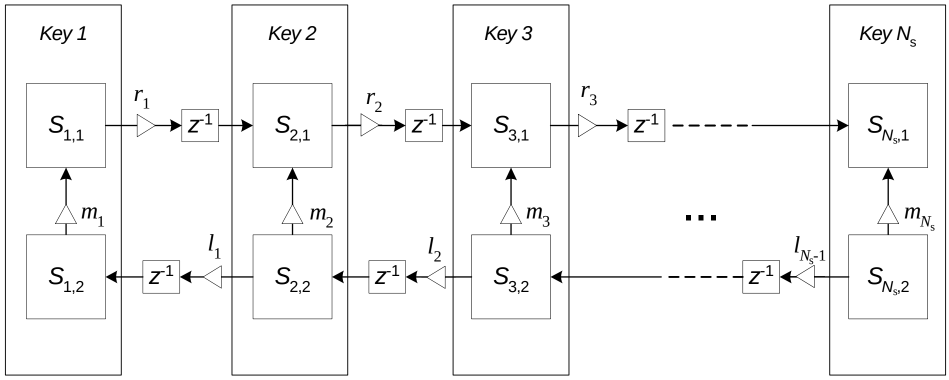

Figure 3.16: *Block diagram of the sympathetic resonances simulation method.* $S_{i,1}$ *denotes the primary string model of key* i*, and* $S_{i,2}$ *denotes the secondary string model.*

addition, the signal is multiplied with coefficient r_i or l_i between two string models. The block diagram of the simulation method is shown in Figure 3.16 and the block diagram of the two string models related to single key is presented in Figure 3.17. The method can be simplified if desired by using constant coefficients, by neglecting the connection from the secondary string to the primary string, and by routing the sympathetic signal through all initialized string models without paying attention to the key order.

A major advantage of this method is that stability is guaranteed as there is no feedback loop in the system. In addition, the simulation can be implemented efficiently, as only undamped strings are needed in the simulation. A piano synthesizer usually has limited polyphony, e.g. 16 or 32, and the synthesizer engine will need to keep track on the strings/keys that are active. Hence, this information is available to be used with the sympathetic resonances simulation. The major difference between the proposed method and the method suggested by Borin et al. [129] is that in the proposed method the sympathetic signal can be controlled for each string model, whereas the latter method produces practically the same sympathetic signal for each string model. On the other hand, the latter method is computationally more efficient, as it does not require double strings per single key.

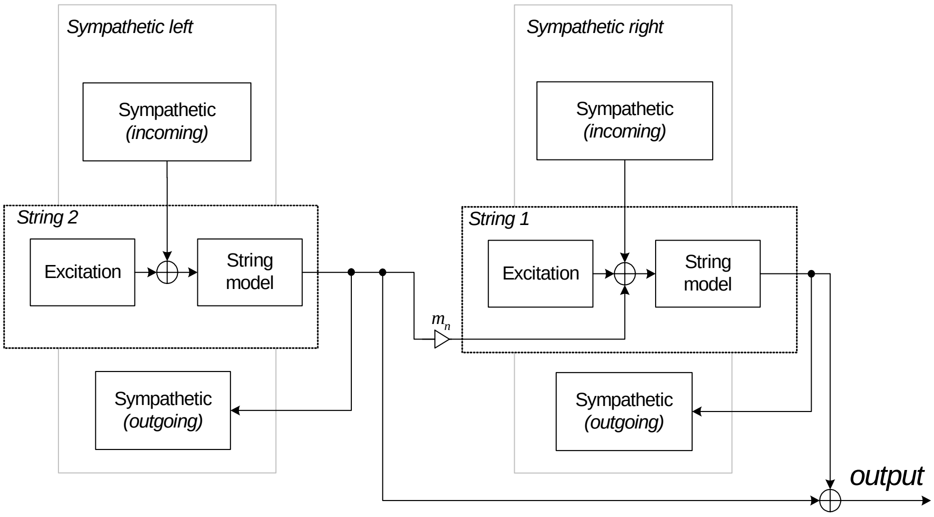

Figure 3.17: *Block diagram of the sympathetic resonances simulation method of a single key.*

Figure 3.18 shows an example of a synthetic piano tone simulated using Matlab with the proposed sympathetic resonance method. In this example, keys E_4 and G_4 are pressed down slowly in the beginning. Then, a loud staccato note is played on key C_2, which causes the undamped string corresponding to the keys E_4 and G_4 to vibrate more loudly, as desired. This suggests that the proposed method is able to simulate the phenomenon. Corresponding sound examples can be found at http://lib.tkk.fi/Diss/2007/isbn9789512290666/.

3.6 Reference implementation with PWGL software

A real-time implementation of the piano synthesis model presented in this thesis is presented in [PVII]. The model is implemented with PWGL software [147, 148, 149, 150] in collaboration with the Sibelius Academy and it does not use any sampled sounds for producing sounds. The software implementation has two goals: to provide evidence that the proposed new methods for piano synthesis can be implemented in real time, and to

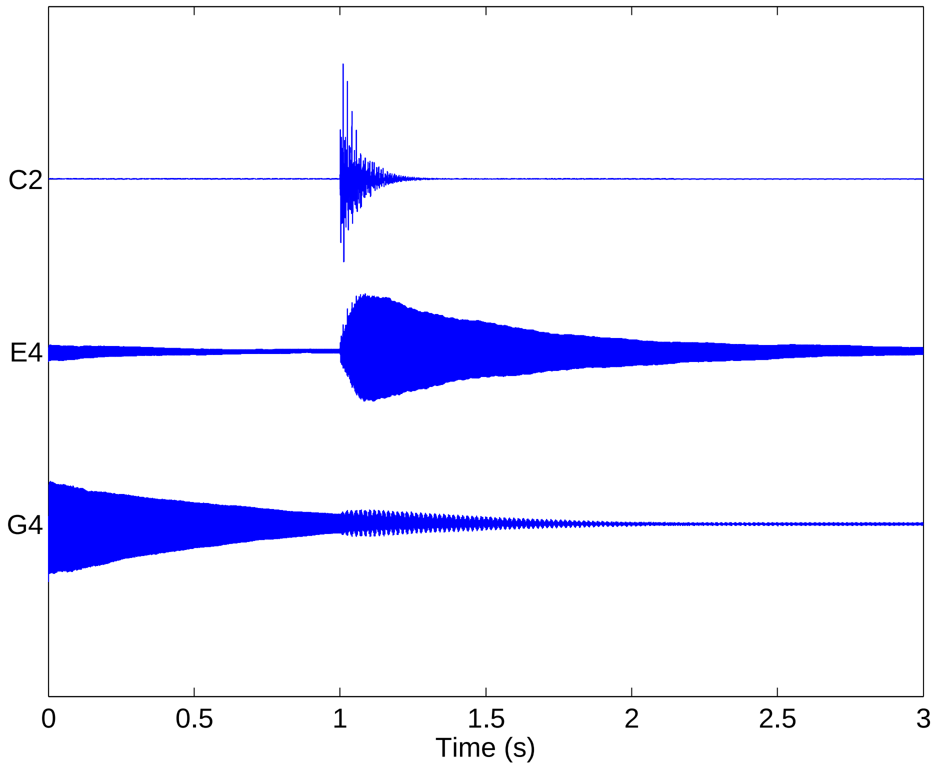

Figure 3.18: *Examples of string model output envelopes of keys* C_2, E_4, *and* G_4, *when the string models are connected with the proposed sympathetic resonance simulation method. In this case, a loud staccato note played on key* C_2 *causing energy flow to the strings corresponding to the other keys, which is seen in the envelopes as an additional string excitation at around one second. The envelopes are scaled in order to emphasize the phenomenon.*

show how the parameter control can be realized in real time. The first goal has been fulfilled as the resulting piano model incorporates all of the proposed methods (the model includes the beating effect simulation proposed in [PIV], and the simplified version of the sympathetic resonance simulation). A screenshot of a single piano string model in PWGL is presented in Figure 3.19 showing the blocks including the excitation model, combined delay line and loss filter block, dispersion filter, tuning filter, combined beating model and knocking tone simulation block, and the connections between the blocks.

The second goal has also been met, as the model includes a functionality where the fundamental frequency and the inharmonicity coefficient value of a single string can be modified in real time. Tuning certain parameters of a specific string is realized with a table, where the parameter values can be changed on the fly. The new internal parameter values for each block corresponding to a certain key are recalculated each time the key is pressed down in the MIDI keyboard. Hence, it is possible to fine-tune e.g. the inharmonicity coefficient value by ear. An example of this is shown in Figure 14 of [PVII].

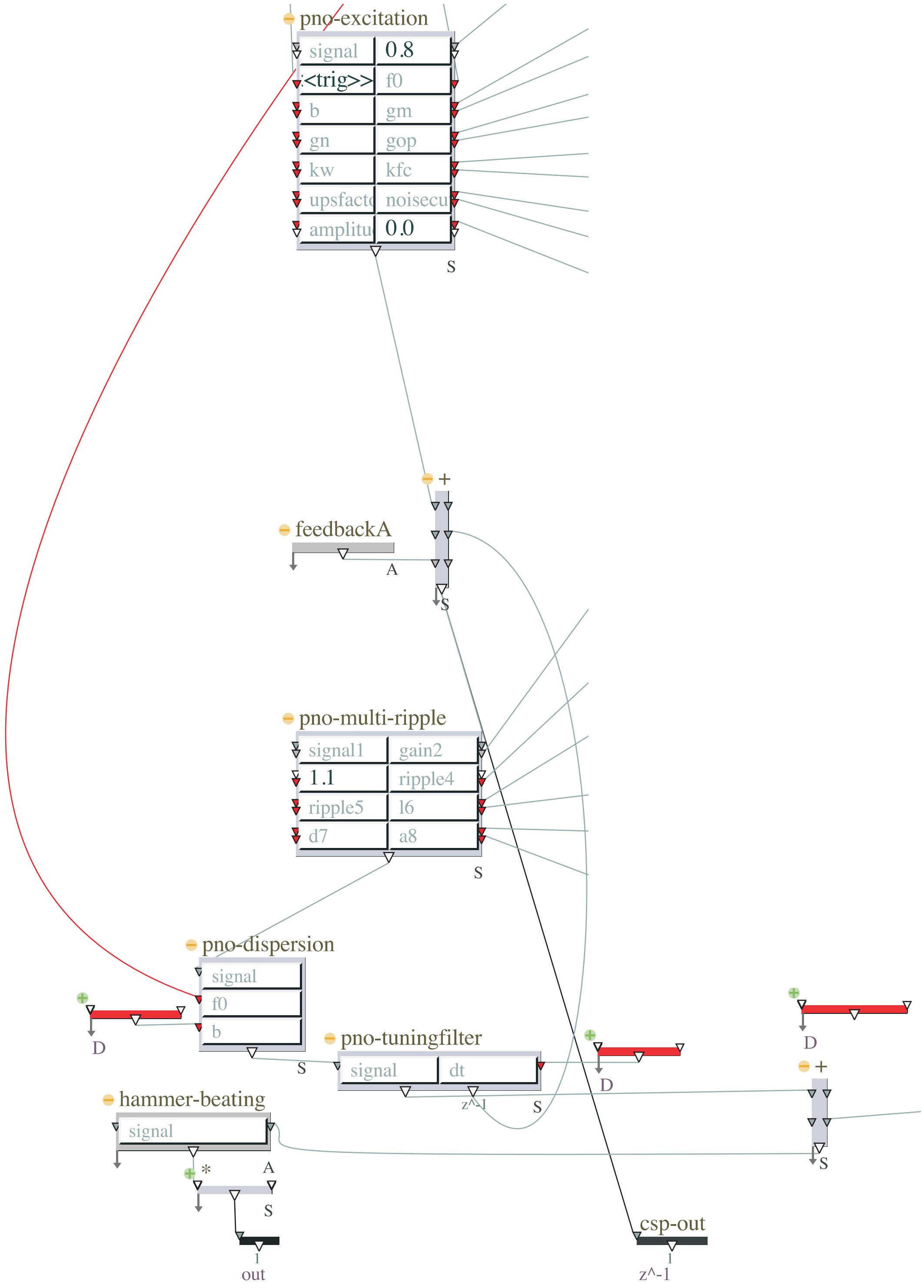

Figure 3.19: *Screenshot of the piano string model implemented in PWGL software (adapted from Figure 13 of [PVII]). The excitation model is denoted as pno-excitation, the combined delay line and loss filter block as pno-multi-ripple, the dispersion filter as pno-dispersion, the tuning filter as pno-tuningfilter, and the combined beating and knocking tone block as hammer-beating.*

4 Conclusions and future research directions

The theme of this thesis has been physics-based modeling of the piano. More specifically, the development of new simulation methods that enable the implementation of an efficient and parametrically controlled piano synthesis model. Work has been done to develop methods for simulating the dispersion phenomenon, the string excitation, the beating effect, and the sympathetic resonances phenomenon. As a result, a novel excitation model has been proposed that is able to excite the string model in the desired way while being able to maintain control of the model through parameters. Moreover, the first closed-form design method for designing dispersion filters has been developed for designing first- and second-order dispersion filters. In addition, an automatic analysis algorithm for estimating the inharmonicity of piano tones has been developed to be used to assist sound synthesis. Furthermore, two new amplitude modulation-based beating effect simulation methods have been suggested. These two methods can produce the desired beating envelope efficiently and the methods can be controlled with parameters in real time. Also, a new method for simulating the sympathetic resonances has been introduced. Finally, these methods have been implemented in a real-time piano synthesis model using the PWGL software proving that the goal for the work has been met.

As for future work, one of the main areas of research relates to the human perception of piano tones, which provides essential information to be used in developing more efficient and perceptually accurate sound synthesis models. At this moment, considering this work, there is need for perceptual information related to dispersion filter design, loss filter design, and excitation model calibration. Another interesting path of study is to investigate how to implement piano synthesis models, which can be scaled in terms of memory requirement and processor power, based on the parametric approach presented in this thesis. This work could be used, for example, in mobile phones and gaming devices. Finally, an important part of the sound synthesis is the calibration of model parameters. The introduced model could be used to develop an automatic calibration method that calibrates the model based on a set of recorded piano tones. Hence, it would enable the simulation of different pianos with the same model by obtaining different parameter sets for each simulated piano.

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