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# Loudness and timbre issues in plucked stringed instruments - analysis, synthesis, and design

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Abstract

This thesis explores acoustical properties of plucked stringed instruments and proposes sound synthesis methods for plucked stringed instruments. In this thesis the investigations of the kantele, the acoustic guitar, the harpsichord, and the guqin provide analysis results, a control algorithm, and two model-based sound synthesis methods in the following way. For a modified design of the kantele, acoustical measurements and formal listening tests support the assumption of increased loudness. The investigation also provides design rules to make a plucked stringed instrument louder. An automatic plucking-point estimation algorithm is proposed for acoustic guitars with an under-saddle pickup. The automatic plucking-point estimation can be used for controlling synthesis algorithms or sound effects. Model-based sound synthesis algorithms are proposed for the harpsichord and the guqin. It was found that the harpsichord exhibits dynamic behavior, albeit limited, when the tangent is pressed down with different speeds. Moreover, the string vibrations and the body response show changes according to the playing level. Hence, the proposed synthesis algorithm is dynamic. The analysis of the guqin reveals the existence of phantom partials in the tones, differences in tones according to the termination technique, and extensive use of flageolet tones and sliding of tones. The model-based sound synthesis algorithm takes these features into account.



Keywords digital signal processing, musical acoustics, model-based sound synthesis, computer music

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Tiivistelmä

Tämä väitöskirja käsittelee näppäiltävien kielisoittimien akustiikkaa ja esittää äänisynteesimenetelmiä näppäiltäville kielisoittimille. Väitöskirjassa esitellään suunnittelusäännöt äänekkäälle kanteleelle, automaattinen menetelmä kitaran näppäilykohdan arviointiin, virtuaalinen cembalo ja äänisynteesimalli kiinalaiselle soittimelle. Suunnittelusäännöt kanteleen äänekkyyden parantamiseksi on kehitetty yhteistyössä soitinrakentaja Jyrki Pölkin kanssa. Sääntöjen pätevyys osoitetaan laskennallisesti, akustisin mittauksin ja kuuntelukokein. Akustiselle kitaralle kehitetty automaattinen näppäyskohdan arviointimenetelmä toimii soittimessa, jossa kontaktimikrofoni on sijoitettu tallan alle. Tätä automaattista menetelmää voidaan käyttää ohjaamaan esimerkiksi äänisynteesiä tai ääniefektiä. Lisäksi väitöstyö esittelee mallipohjaiset äänisynteesimenetelmät cembalolle ja historialliselle kiinalaiselle kielisoittimelle, jonka nimi on guqin. Tutkimus osoittaa, että cembalossa esiintyy dynaamista vaihtelua, kun kosketinta painetaan eri nopeuksilla. Lisäksi kielien värähtelyssä ja kaikukopan vasteessa esiintyy eroja soittotason suhteen. Tästä johtuen kehitetty äänisynteesimalli on dynaaminen, jonka lisäksi mallilla voidaan tuottaa cembalomusiikkia, jossa on suuremmat äänenvoimakkuusvaihtelut kuin todellisuudessa. Guqinille analyysitulokset osoittavat, että soittimessa esiintyy haamuääneksiä, eroja äänessä kielen sormitustavasta riippuen ja laaja-alaista huilu- sekä liu’utusäänten käyttöä. Esitetty synteesimalli ottaa nämä ominaisuudet huomioon.



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# Preface

This work was carried out in the Laboratory of Acoustics and Audio Signal Processing, at Helsinki University of Technology (TKK), Finland during the period between 2003 to 2006. Since day one the colleagues at the laboratory have given an incredible amount of support and help. In addition, I have had the privilege to work with skilful people from an exceptionally large range of disciplines such as, instrument builders, musicians, a psychologist, software developers, and naturally engineers. This preface aims at listing most of the important people involved and who I am ever grateful to. For better or worse, this whole trip changed my life and I'll never be the same again.

First of all, I would like to thank my supervisor Prof. Vesa Välimäki for his supervision and support throughout these years. It enabled me to bring the work to a completion and you this thesis. I also want to express my gratitude to Prof. Matti Karjalainen who has been very helpful regarding this work and for showing his warm attitude. What is more, I've been fortunate enough to have experienced the tremendous support, extreme high level of professionalism, musical skills, and positivity of collaborator Dr. Cumhur Erkut. I'm also thankful to have co-operated with Dr. Mikael Laurson and Mr. Jyrki Pölkki, since they are superb professionals and great characters. In addition, I would like to thank my other co-authors for helping me in giving content to this work, Prof. Marc Leman, Ms. Henbing Li, Mr. Jyri Pakarinen, Mr. Jonte Knif, and Mr. Jaakko Siiskonen. All the co-authors helped me greatly with their comments, ideas, and positive feedback. Furthermore, I would like to thank the pre-examiners of my thesis, Prof. Anssi Klapuri (Tampere Univ. of Tech., Finland), and Associate Prof. Stefania Serafin (Aalborg Univ. Copenhagen, Denmark), for their comments that helped to bring coherence to the text. I'm also thankful for the gentlemen involved in the off-side publication projects Mr. Mika Kuuskankare and Mr. Miikka Tikander.

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A special thank you goes also to Luis Costa who is my most avid reader, as he has been the native English speaker who has corrected the grammar etc. of this thesis and four of my articles.

The Aku Lab has been an amazing environment to work in and I'm sure I could've not experience anything like it anywhere else. Ms. Lea Söderman will always be my favorite secretary, through her professional skills and her cheerful persona. Aki Härmä was a great model for me and I want to thank his support and encouragement to continue his and Unto Laine's work on the SPÄNK course (Digital Signal Processors and Audio Signal Processing), and all the amazing co-assistants Miikka Tikander, Julia Jakka, Markku Ursin and assistants Juha Merimaa, Antti Huovilainen, Toni Liitola, Hannu Pulakka, and Matti Airas. I enjoyed and appreciate the company and help of my room mate Tom Bäckström, people around Siberia, Tuomas Paatero, Patty Huang, Jussi Pekonen, Jukka Rauhala, Seppo Fagerlund...; all the Ladies room versions, Riitta Väänänen, Ville Pulkki, Kalle Palomäki...; The Moscow area, Mara Rahkila, Hynde,...; The Tsetsenia crew, Toni Hirvonen, Carlo Magi, Petri Korhonen; Miikka's brother-in-arms Toomas Aaltosaar, The ladies of Siberia, Laura Lehto, and Heidi-Maria Lehtonen, and their Swedish reinforcements Eva Björkner, and Laura Enflo. I also want to thank Hanna Järveläinen for her friendship and support in and outside the lab.

This preface is already ridiculously long and I'm not even finished yet. There's a vast realm of people outside the work community which have kept me intact and inspired. I would like to give special acknowledgments and a hats-off-bow to: long time friends, Lauri Halme, Marko Piispa, Panu Kaukinen, Harry Kuusela, Ville Palkosaari, and Miika Valve, and ex-room mate Jussi Lampiselkä. My friends introduced through music, but not only; Antero Kuisma ja Suojatyöpiakka, Janne Toivanen, Petri Naukkarinen, Perttu Hämäläinen, Jussi Koski, and Kristian Sahldstet; Kitarakerho, Petteri Kekäläinen; Bottle Consotrium; Jaakko Prättälä, Mikko Ojanen, and Anna Dantchev; and Miikka Järvinen.

Finally, I'm grateful for the endless support and love I have received from my mother and my sister.

This work is dedicated to my father Markku Penttinen.

Henri Penttinen, Espoo, Finland, 13.9.2006

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# List of Publications

P1 Penttinen H., Erkut, C., Pölkki, J., Välimäki, V., and Karjalainen, M. “*Design and analysis of a modified kantele with increased loudness,*” Acta Acustica united with Acustica, vol. 91, pp. 261–268, 2005. Special issue on Musical Acoustics.

P2 Penttinen, H., and Välimäki, V., ”*A time-domain approach to estimating the plucking point of guitar tones obtained with an under-saddle pickup,*” Applied Acoustics vol. 65, no. 12, pp. 1207–1220, December 2004.

P3 Penttinen, H., Siiskonen, J., and Välimäki, V., ”*Acoustic guitar plucking point estimation in real time,*” in Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing, vol. 3, pp. 209–212, Philadelphia, PA, USA, March 19-23, 2005.

P4 Välimäki, V., Penttinen, H., Knif, J., Laurson, M., and Erkut, C., ”*Sound synthesis of the harpsichord using a computationally efficient physical model,*” EURASIP Journal on Applied Signal Processing, vol. 2004, no. 7, pp. 934–948, 15 June 2004. Special Issue on Model-Based Sound Synthesis.

P5 Penttinen, H., ”*On the dynamics of the harpsichord and its synthesis,*” Accepted for publication in Proc. of the Int. Conf. on Digital Audio Effects DAFx06, September 18-20, Montreal, Canada, 2006.

P6 Penttinen, H., Pakarinen, J., Välimäki, V., Laurson, M., Li, H., and Leman, M., ”*Model-based sound synthesis of the guqin,*” Tech. Rep. 78, Lab. of Acoustics and Audio Signal Processing, Helsinki University of Technology, Sept. 2006, (Submitted to the Journal of the Acoustical Society of America on May the 5th 2006).

# Author's Contributions

## Publication [P1]

The author made the acoustical measurements and reported these results in collaboration with the second author by being the corresponding person (Secs. 1, 2, 3, and 5). Co-author #3 also provided input to Sec. 1. The author conducted the listening tests and reported them in Sec. 4. The initial design rules were suggested by instrument builder Jyrki Pölkki.

## Publication [P2]

The author conducted the experiments and reported the greater part of the results discussed in this article. The method was developed in collaboration with the co-author, who provided input to Sec. 5. The author also prepared the companion webpage.

## Publication [P3]

The author developed the improvements of the algorithm, coded the simulations, and initiated the tests and supervised the coding of the real-time system. The author wrote the publication and prepared the companion webpage.

## Publication [P4]

The author conducted the acoustical measurements and recordings of the harpsichord in co-operation with the third author. The author created the basic excitation sample database for the harpsichord synthesizer, which the co-authors extended with their design tools. The author was responsible for the body model design and calibration, and was the main contributor in Sec. 2, 3.4, and 4.5. He also contributed to Secs. 3.2, 3.3, and 4.1. In addition, the author contributed to the overall polishing and commenting of the paper.

## Publication [P5]

The author was responsible for this research. Discussions with colleague Jonte Knif supported the initial planning of the work.

## Publication [P6]

The author was responsible for the conducted recordings and measurements of the instrument. The author created the sound synthesis model in co-operation with the second and third authors. The acoustical measurements and the analysis of the instrument were done by the author, except for the friction sound and energy compensation of the string. The author was the main writer of sections III A and B, intro to IV, IV A.3, B, C.1, and C.2, and VI. Section I was written with co-authors #3 and #6 and Sec. II was written with co-author #5, respectively. In other words, the author was mainly responsible for writing the article except for Secs. III C, IV A.1, A.2, C.3, and V. The author also prepared the companion webpage.

# List of Abbreviations

| | |
|---|---|
| CWS | Commuted Waveguide Synthesis |
| DSP | Digital Signal Processing |
| DWG | Digital Waveguides |
| ENP | Expressive Notation Package |
| FD | Finite-Difference |
| FDTD | Finite-Difference Time-Domain |
| FIR | Finite Impulse Response |
| IIR | Infinite Impulse Response |
| LTI | Linear and Time Invariant |
| PD | Pure Data |
| SDL | Single-Delay Loop |
| STK | Synthesis Tool Kit |
| TMDF | Tension Modulation Driving Force |

# List of Symbols

| | |
|---|---|
| $A_{\mathrm{d}}(z)$ | Allpass filter |
| $A_{\mathrm{n}}$ | Initial amplitude of mode $n$ |
| $a$ | Loop coefficient in the loop filter |
| $c_{\mathrm{l}}$ | Velocity of longitudinal wavefront |
| $c_{\mathrm{t}}$ | Velocity of transversal wavefront |
| $E$ | Young's modulus |
| $F(f)$ | Fourier transform of the force exerted to the support |
| $F(z)$ | Fractional delay filter |
| $f(x,t)$ | Force distribution |
| $f_0$ | Fundamental frequency |
| $f_1$ and $f_2$ | Waves traveling to left and right, respectively |
| $f_{\mathrm{n}}$ | Frequency of mode $n$ |
| $\varphi_n$ | Initial phase of mode $n$ |
| $g$ | Gain coefficient in the loop filter |
| $H_{\mathrm{LF}}(z)$ | Loop filter transfer function |
| $L$ | Length of the string |
| $m$ | Mass of the string |
| $\mu$ | Linear mass density |
| $n$ | Mode number |
| $r$ | String radius |
| $S$ | Cross-sectional area of the string |
| $S(z)$ | Transfer function of the DWG string model |
| $T$ | Tension applied to the string |
| $t$ | Time |
| $\tau_n$ | Decay time of mode $n$ |
| $\tau_{\mathrm{air}}$ | Viscous decay time |
| $V(f)$ | Fourier transform of the velocity exerted to the support |
| $x$ | Coordinate a long the string |

| | |
|---|---|
| $Y(f)$ | Fourier transform of the mechanical admittance at the bridge |
| $y$ | Displacement |
| $z^{-L_1}$ | Delay line of integer length $L_1$ |
| $\frac{\partial y}{\partial x}$ | Spatial derivative |
| $\frac{\partial y}{\partial t}$ | Temporal derivative |

# Chapter 1

# Introduction

## 1.1 Background

This thesis deals with analysis of musical instruments and sound synthesis methods to mimic the sounds of plucked stringed instruments. Analysis of the instruments falls typically under the category of musical acoustics, while sound synthesis is a part of music technology. Also, as in this thesis, analysis results can be used directly for purposes of music technology. A sound synthesis method that imitates, on a computer, the behavior and sound production mechanism of a sound source is called physical modeling synthesis or model-based sound synthesis [1, 2, 3, 4, 5, 6, 7, 8, 9].

The history of plucked string instruments started about four thousand years ago in Mesopotamia, situated in modern day Iraq [10]. The construction of stringed instruments has evolved tremendously over the years, and plucked stringed instruments are nowadays practically used in every corner of the globe. Some musical instruments are fine examples of constructions with complicated machinery, such as the piano and the harpsichord. Also, the violin is an example of a sophisticated instrument. The violin and the piano have played a crucial role in the development of western music. What is more, they both have a significant role in the progression of musical acoustics [11, 12, 13]. They have played an important role in acoustical findings and investigations regarding phenomena such as inharmonicity [14, 11], phantom partials and longitudinal vibrations [15, 16], string coupling [17], Helmholtz motion and wolf tones [12], and body modeling [18]. The violin has also been a part of virtual reconstruction by eliminating both the strings and the resonating body [19]. Plucked stringed instruments are also part of this trajectory, and, for example the guitar has been investigated thoroughly [20].

Besides the general development of musical instruments over the years, the trend of building louder instruments has been on going for a few hundred years now. This is to meet the requirements of larger audiences and concert halls. The evolvement of keyboard instruments, or the emergence of the piano, is a prime example of this kind of a progression: clavichord ⇒ harpsichord ⇒ piano [20]. The acoustic guitar underwent a stepwise jump in the treatment of luthier de Torres [20]. Similarly, the modern flute is an example where a single instrument builder, Mr. Boehm, made significant changes. The violin has also experienced many changes over the years, i.e., a violin built today differs from the ones built by Antonio Stradivari. That is to say, today the neck is longer, the strings are under higher tension, and the bass bar, or sound bar, is tuned better than during the baroque era [20].

In the field of sound synthesis, the first physical modeling scheme was proposed in the early 1970s by Hiller and Ruiz [21, 22], by using the finite-difference time-domain (FDTD) approximation of the wave equation. The first real-time system was the CORDIS system presented in the early 1980s [23, 24]. The first commercial synthesizer, the Yamaha VL-1, applying physical modeling or model-based synthesis was introduced in 1994 [25]. Consequently, the field of sound synthesis is still relatively young compared, for example, to the history of acoustical instruments. The research around sound synthesis, and especially model-based synthesis has been active during recent years (see for example recent doctoral theses [26, 27, 28, 29, 30, 31, 32, 33, 34]). The discussion on sound synthesis methods will continue in Section 3.

Control issues are also a crucial part of sound synthesis and have been discussed, e.g., in [35, 36, 37]. Software environments as well are important for sound synthesis and research purposes. Such applications have appeared: e.g., the Synthesis Tool Kit (STK) [38], Pure Data (PD) [39, 40] and physical modeling extensions for it [41, 42], the BlockCompiler [43], and the PWGLSynth [44].

The usage of physical models in music has mostly occurred via modern and experimental composers [45] and this widespread use is hopefully just around the corner. The concept physics in physical models can partly scare some composers and restrain them to use physical sound synthesis models. On the other hand, even if physical models can seem complex the control parameters involved have strong equivalence with a real instrument. Therefore, the control parameters should be more familiar to the users, but so far it is not yet the case. This thesis deals with the issues for making stringed instruments louder, proposes a method for control, and attempts to narrow down the gap between academia and musicians by for example using the PWGLSynth [44] for implementations of the sound synthesis algorithms it proposes.

## 1.2 Scope of the thesis

The scope of this thesis is analysis and model-based sound synthesis of plucked stringed instruments. The instruments used in this thesis are the harpsichord, the kantele, the guqin, and the acoustic guitar. Acoustical measurements and analysis have been performed for all the instruments. The output and goals of these analysis procedures vary. The main focus areas are loudness and timbral investigations, including plucking point identification and model-based sound synthesis.

Acoustical measurements have been performed in the form of recording of tones and playing with microphones, vibrations of the body have been measured with microphones and a laser vibrometer. Also, formal listening tests have been executed to gain knowledge of the perceived loudness of the kantele and the harpsichord. When applied to model-based synthesis, on one hand, the measurement data have been used to understand the properties of the instrument. On the other hand, the measurement data have been used to calibrate the synthesis models. Thereafter, the measurement data are compared, by signal analysis means, with the output of the synthesis to verify the legitimacy of the synthesis algorithm. Calibration is based on methods introduced and discussed in [46, 47, 48]. Extensions to these calibration routines in the publications of this thesis are given and used regarding inharmonic inverse filtering [P4 and P6], flageolet tone synthesis [P6], and synthesis of phantom partials [P6].

The model-based sound synthesis approach taken in this thesis is physically oriented. The sound synthesis models are designed with the knowledge that they will be used and implemented in a real-time synthesis environment. The synthesis environment encompasses the synthesis engine PWGLSynth [44], which is a visual synthesis language, and the Expressive Notation Package (ENP) [49], which controls the PWGLSynth. Recently, an interface has been designed for the PWGLSynth for real-time control purposes [50].

Models for loudness have been developed and they are evolving towards modeling of time-varying properties [51, 52, 53]. However, the models are very recent and are not yet at a satisfactory level to replace formal listening tests with human subjects. This applies especially to the case of a vibrating string, since the behavior of a vibrating string is highly non-stationary as it exhibits timbre changes, initial pitch drift, non-linear phenomena, which all occur as functions of time. Since this thesis concentrates on analysis and synthesis of musical instruments, the development of perceptual loudness models is outside the scope of this thesis.

The sound synthesis algorithms proposed in this thesis can be viewed as virtual instruments that mimic and approximate acoustic instruments. This thesis touches on some of the control issues regarding sound synthesizers, e.g., in [P2] and [P3], and in [P4] - [P6], in terms of musical control of an expressive sound synthesizer with timbre, pitch, and loudness properties. On the other hand, there is the world of pure virtual instruments, where a synthesis engine is controlled with a digital interface in a way where the objective is to create completely new instruments [54, 55, 56]. These instruments and their control issues [57, 58, 59] are beyond the scope of this thesis.

## 1.3 Content of the thesis

This doctoral thesis consists of six articles and an introduction. The articles investigate the behavior of plucked string instruments through analysis, synthesis, and design, with an emphasis on loudness[1] and timbral[2] issues. The articles in this thesis cover a wide range of issues related to plucked string instruments, some of the issues overlapping with each other. These overlaps are highlighted by looking at the content of the articles from different view points.

Analysis has been performed in all the publications of this thesis, but in [P1], [P2], and [P3] the emphasis is particularly on acoustical measurements and signal analysis. More specifically, in [P1] rules based on the ideas of instrument builder Jyrki Pölkki for making a plucked string instrument louder are introduced. These rules have been implemented and verified in the case of the kantele by means of acoustical measurements and subjective listening tests. Furthermore, articles [P2] and [P3] follow the development of an automatic plucking point algorithm. Sound synthesis has a more significant role in articles [P4], [P5], and [P6], so that [P4] and [P5] discuss the sound synthesis and analysis of the harpsichord and [P6] tackles the Chinese instrument called the guqin. Loudness issues are discussed in publications [P1] and [P5], where signal analysis has been conducted and listening tests have been performed to investigate the perception of the sound of these instruments. Timbral issues are present in all the articles in one way or another. Moreover, the body or soundboard of a musical instrument has a major effect on its timbre. Body modeling methods are proposed in [P4], [P5], and [P6].

[1] loudness - the attribute of a sound that determines the magnitude of the auditory sensation produced and that primarily depends on the amplitude of the sound wave involved. (Merriam-Webster Online, URL: http://www.m-w.com/, 2006.)

[2] timbre - that quality which distinguishes two sounds with the same pitch, loudness, and duration. (as defined by the Acoustical Society of America, ASA)

# Chapter 2

# Acoustics and analysis of plucked string instruments

Next, the acoustics of plucked stringed instruments are discussed relating prior and ongoing research to this thesis. To understand the acoustics of plucked stringed instruments, it is beneficial to look at the system as separate sub-blocks, as illustrated in Fig. 2.1. Consequently, this also helps to discuss the synthesis of the instruments. Good sources for covering the basic concepts and the research field in general are [60] and [20], whereas reference [61] gives an in-depth technical review and [62] discusses issues related to the physics and numerical simulations of stringed instruments.

The divided system shown in Fig. 2.1 works as follows. The system, i.e., the stringed instrument, is excited by plucking (exciter). In other words, the player displaces the string by applying a force distribution $f(x,t)$ on the string, where $x$ is the coordinate along the string and $t$ denotes time. The string is then released and the system starts to vibrate. The vibration energy of the string is then distributed via the bridge to the soundboard and from there to the cavity of the instrument and the surrounding air. These distribution steps are separated from each other in Fig. 2.1, but are connected with bidirectional arrows. This means that the systems are coupled to each other and interact as a function of time. As will be discussed later, these couplings result in characteristics that help to explain the properties of stringed instruments. Figure 2.1(b) shows a generalized plucked stringed instrument from where most instruments of the family can be drawn. Optional parts, such as the neck, frets, and soundhole, are shown with a dashed line. In the guitar the saddle is a part that rests on top of the bridge and guides the strings, whereas frets are at right-angles to the strings and placed in small grooves on the neck or fingerboard. In the following the separate sub-blocks are discussed in more detail.

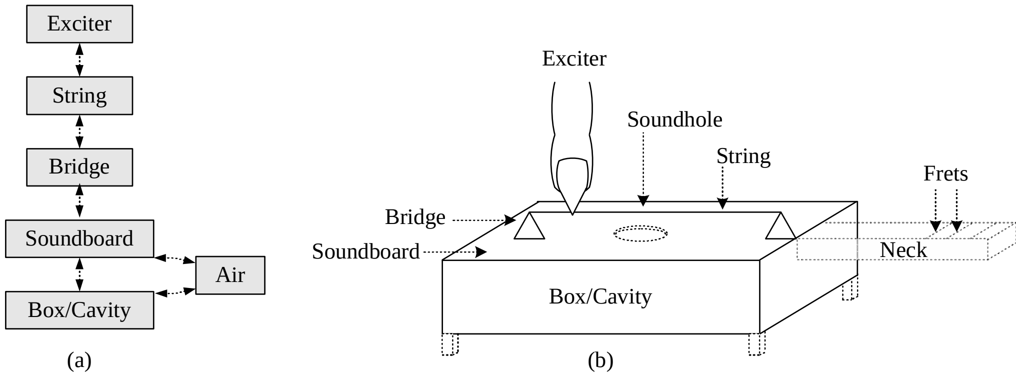


Figure 2.1: Two block diagrams of a plucked stringed instrument. (a) The functional sub-blocks and (b) a generalized plucked stringed instrument with optional parts shown with a dashed line.

## 2.1 Vibrating string

Research around a vibrating string is not in its adolescence anymore. Pythagora has been reported to have noticed to the pleasing sound two segments of a stretched string produced when the ratio of the segments is simple, such as 2:1, 3:1, 4:1, and so on [20][1]. These are examples of the normal modes of a string fixed at its ends. Later, it has been revealed that the normal modes depend upon the mass $m$ of the string, its length $L$, the tension $T$ applied to it, and the end conditions [20]. By expressing the linear mass density $\mu$ of a stretched string, the familiar wave equation for transverse waves in a vibrating string, can be expressed as

$$f(x,t) = \frac{\partial^2 y}{\partial t^2} - \frac{T}{\mu}\frac{\partial^2 y}{\partial x^2} = \frac{\partial^2 y}{\partial t^2} - c_{\mathrm{t}}^2 \frac{\partial^2 y}{\partial x^2}, \tag{2.1}$$

where $c_{\mathrm{t}}$ is the speed of the propagating transverse wave, $t$ is time, $x$ the coordinate along the string, and $y$ is the displacement of the string. $\frac{\partial y}{\partial x}$ and $\frac{\partial y}{\partial t}$ are the spatial and the temporal derivatives, respectively. Equation 2.1 is a simplification of what happens in a real string but, for example reveals aspects of how a harmonic, lossless, and flexible string behaves. For a plucked string $f(x,t)$ becomes zero after the release of the string. As expressed in Eq. 2.1 the transversal speed of the wave is $c_{\mathrm{t}} = \sqrt{\frac{T}{\mu}}$. Similarly, the motion of the longitudinal wave can be expressed like in Eq. 2.1, but $c_{\mathrm{t}}$ is replaced by the longitudinal speed

$$c_{\mathrm{l}} = \sqrt{\frac{ES}{\mu}}, \tag{2.2}$$

[1]The ratio 3:2 mentioned in [20] is not a normal mode that an individual string exhibits, but rather a ratio that two strings or two parts of a string can exhibit.

where $E$ is Young's modulus and $S$ is the cross-sectional area of the string.

The traveling wave solution (to Eq. 2.1, when $f(x,t) = 0$) first presented by d'Alembert (1717-1783), can be formulated as

$$y = f_1(ct - x) + f_2(ct + x), \tag{2.3}$$

where functions $f_1$ and $f_2$ represent the waves traveling to the right and left, respectively. In an ideal case for a string, Eq. 2.3 tells us that the vibrations of the string constitute of two pulses traveling in opposite directions. The traveling waves reflect from the end terminations and with $y$ = 0 for a fixed end when $x$ = 0, Eq. 2.3 becomes

$$y = 0 = f_1(ct - 0) + f_2(ct + 0), \tag{2.4}$$

and furthermore

$$f_1(ct) = -f_2(ct). \tag{2.5}$$

This means that a positive pulse, or an up pulse, reflects as a negative pulse. The knowledge of the traveling wave decomposition and the speed of the transversal wave as a function of frequency were crucial for formulating and designing the automatic plucking-point estimation algorithm discussed in [P2] and [P3].

The general solution for the equation of a freely vibrating, lossy, and rigidly supported string that is transversely displaced at a position $0 \leqslant x \leqslant L$, i.e., the plucking point is $x$ and time $t \geqslant 0$ can be stated as

$$y(x,t) = \sum_{n=1}^{\infty} A_n \cos(2\pi f_n + \varphi_n) e^{-t/\tau_n} \sin(\frac{n\pi x}{L}), \tag{2.6}$$

where $f_n$ is the frequency, $\tau_n$ is the decay time, $A_n$ is the initial amplitude, and $\varphi_n$ is the initial phase of mode $n$, and $L$ is the length of the string. The $e^{-t/\tau_n}$ term represents the losses. From Eq. 2.6, by using the Fourier analysis, $A_n$ can be solved for fixed boundary conditions so that [63]

$$f_n = \sqrt{\frac{T}{\mu}} \frac{n}{2L} \tag{2.7}$$

and

$$A_n = \sqrt{\frac{2}{L}} \sin \frac{n\pi x}{L}. \tag{2.8}$$

With these solutions one can look at the effect the plucking point has in the frequency domain. This is the typical and most obvious way to look at the consequences of the plucking point, also for automatic estimation purposes as in [64, 65, 66], because it reflects how the plucking point effect is auditorily perceived.

To include dispersion, energy dissipation, and the transfer of energy from the string to the rest of the system Eq. 2.1 has to be altered. Naturally, the solutions change accordingly. The changes and mathematical solutions are discussed in detail, for example in [67].

### 2.1.1 Strings loose energy

After the initial release of the string the vibrations start to decay. There are three main causes for these losses: air damping, internal damping, and energy losses through the supports [20]. These damping mechanisms are frequency dependent and their effect varies in relation to each other and the rest of the vibrating system [68].

Air damping has to do with the fact that a vibrating string radiates poorly. This is because the string acts as dipole radiator that produces a compression (dense) in front and a rarefaction (sparse) behind as it moves. The radius is so small that these fronts effectively cancel each other out[2]. However, this means that the string interacts with the surrounding air. Hence, under some conditions viscous flow of air around the moving string can cause considerable damping [20]. The effect of viscous drag causes the mode frequency to lower very slightly and an exponential decay in amplitude. The viscous decay time $\tau_{\mathrm{air}}$ is proportional to the string density but depends on the string radius $r$ and frequency: at low frequencies as $\tau \propto \mu r^2$ and at high frequencies as $\tau \propto \mu r/\sqrt{f}$.

Internal damping is prominently characterized by the Young's modulus of the string. Moreover, all strings show an elastic behavior where when a stress is applied, immediately a strain occurs that increases slightly with time. This chain of events can be presented with a complex Young's modulus [20, 69]

$$E = E_1 + iE_2. \tag{2.9}$$

This equation represents the viscoelastic losses, while the thermoelastic losses are described with the same formula but with a different ratio of $E_1$ and $E_2$. Viscoelastic losses affect mainly the high frequencies and thermoelastic losses have a resonant frequency at a lower frequency region.

Energy loss through the supports is introduced via the mechanical characteristics of the supports. The mechanical characteristics of the supports, i.e., the way the string and the bridge are in contact with each other, is usually described as the mechanical

[2] Somewhat similar behavior occurs during the radiation from the soundboard of a traditional kantele. This is one of the things design rule III proposed in [P1] tries to prevent.

admittance (reciprocal of impedance)

$$Y(f) = \frac{F(f)}{V(f)}, \tag{2.10}$$

where $F(f)$ and $V(f)$ are the Fourier transforms of the force exerted on the support and the speed at the support, respectively. The damping is described by the real part of $Y$ and the imaginary part changes the effective length of the string, respectively [63]. Measurement techniques for measuring the admittance for the guitar are discussed in [70] and for the violin in [71, 72]. Measuring the admittance functions of the kanteles investigated in [P1] had a crucial role in the analytical comparisons of the input power of the instruments.

On the whole, the different damping mechanisms contribute to the decay time in the following manner

$$\frac{1}{\tau} = \frac{1}{\tau_{\mathrm{air}}} + \frac{1}{\tau_{\mathrm{int}}} + \frac{1}{\tau_{\mathrm{sup}}}. \tag{2.11}$$

In the better part of musical instruments, the rate for energy transfer from the string to the bridge and soundboard is slow. In other words, usually air and internal damping are dominant loss mechanisms. In more detail, in metal strings air viscosity often determines the decay time for upper partials as $1/f$ and in gut or nylon strings the internal damping prevails with a $1/f^2$ trend [20].

### 2.1.2 Strings interact through the bridge

Vibrating strings interact within the string and with other strings, both interactions occur through the bridge. The interactions within the string occur between the three orthogonal directions (longitudinal, horizontal, and vertical), for example, either one of the transverse directions (horizontal or vertical) are coupled with the longitudinal vibrations [16]. Furthermore, the admittance function given in Eq. 2.10 can be generalized to an admittance matrix that describes the behavior of the bridge via three orthogonal directions [73].

Real strings exhibit two-stage decay and amplitude beating of decaying partials due to differences between admittance functions of the horizontal and the vertical directions [17]. In other words, if the vibrational modes of the string have slightly different frequencies in the horizontal from the vertical direction, complicated non-exponential decay patterns appear.

Some harmonics have a very low amplitude in the spectrum of a string due to the plucking point. Generation of missing or weak harmonics occur if the admittance function in at least one direction has a finite value, i.e., $Y \neq 0$ in any direction [74]. In other words, if a harmonic is missing due to the plucking point it is possible that it regains energy. Moreover, when $Y \neq 0$ the tension modulated driving force (TMDF) is able to couple back to the string [75].

Typically, analytical treatments concentrate on single strings. However, many musical instruments have more than one string, which implies that energy will be transferred via the bridge to the other strings that can start to vibrate. When a string sets a neighboring string, which has not been excited directly, into motion, the vibrations are called sympathetic vibrations. Sympathetic vibrations have been discussed in [17, 76, 77].

### 2.1.3 Strings are nonlinear

The wave equation in Eq. 2.1 is a linear approximation of string behavior. This approximation assumes that the length of the string does not change during vibration. This, however, occurs and affects also the tension of the string, and in effect, causes tension modulation. The first analytical and experimental investigations of nonlinear string vibrations are reported in [78, 79] during the 1940's. Later comprehension of the issue has widened, see for example [80] for an overview. Here, a few related issues are mentioned and a short discussion on the coupling between transversal and longitudinal motion is given.

A fundamental property of all strings, the initial pitch glide, has been proven to be caused by tension modulation [78, 63]. When a string is excited with continuous sinusoidal force that has a frequency close to the fundamental of the string it has been shown that tension modulation also causes whirling motion[3] in the string [82, 81], coupling between different modes and planes of motion [82, 74], and amplitude jumps [83].

The longitudinal motion of a string has been a recent and active subject of research. The effect of the longitudinal vibrations on tone and string design in pianos was pointed out by Conklin [84]. Also, it was found that the amplitude of the longitudinal vibrations is a nonlinear function of the amplitude of the transverse vibrations [85]. Nakamura and Naganuma [86] found a second set of partials with one fourth of the inharmonicity of the main set of partials. Conklin later named these phantom partials [87] and still later explained that the generation occurred through nonlinear mixing [15], apparently without being aware, at the time, of [86]. Also, Woodhouse reported of the same phantom partial set [88]. Recently, Bank and Sujbert aggregated these reports and provided a theoretical explanation for the phenomenon [16]. They explain that the generation of phantom partials and longitudinal free modes arise from the coupling of transverse vibrations to the longitudinal polarization.

[3]Whirling motion - Such motion corresponds to each point of the string moving in phase in an elliptical orbit about the equilibrium position [81].

In other words, the second set of partials found by [86] and named by Conklin as phantom partials[87] exist due to the coupling between longitudinal vibrations and transverse vibrations. In [P6] a linearized model for synthesizing the phantom partials is given.

## 2.2 Vibrating body and radiation

The string is a poor radiator, as mentioned above. Therefore, an acoustic amplifier, a wooden box and/or a plate of some kind, is attached to the stringed instrument. Multitude of practical implementations for this amplification are evident from the wide range of existing musical instruments. The research tradition around the acoustic guitar, the violin, and the piano is wide [20]. For the harpsichord the behavior of soundboards, and air and structural modes have been discussed in [89, 90, 91]. The acoustics of the kantele has been addressed in [92, 93, 94].

To a good approximation the soundboard and body can be considered to behave as a linear and time invariant (LTI) system, as suggested in Refs [18, 95, 96] and by sound synthesis algorithms that have a body model of some kind. An opposite opinion is given in [97] based on experimental data, but this opinion is mentioned as a side remark since no supporting formulation has yet appeared and there is a possibility that the amplitude region where the measurements were conducted cannot force the system into the nonlinear region. On the other hand, for example the top-plate in the tanbur [98] could produce nonlinear behavior due to its curved form.

Typically, at low frequencies the body response of an instrument tends to have a few distinct modes, which are very clear, for example in the acoustic guitar [20]. At higher frequencies the mode density increases and modes overlap heavily with each other. Following the path from the plucking event after the string has interacted with the admittance of the bridge the motion is filtered by a frequency dependent radiation function. In [P1] the radiation characteristics, at a point one meter above the instrument, of two designs of the kantele are compared with each other.

# Chapter 3

# Model-based sound synthesis of plucked string instruments

## 3.1 Model-based sound synthesis

Model-based algorithms aim to model the behavior of the sound source, in this case, the musical instrument at hand. These mathematical formulations produce a discrete output that can be implemented on a computer. When a synthesis algorithm is built this way, based on the physics, its output should resemble the original instrument. The resemblance is not perfect but is typically a good starting point for development. In addition to the resemblance between the sounds, the model-based algorithm possesses resemblances to the real world. This is beneficial in the sense that when a parameter, e.g., the length of the string, of the synthesis model is changed it has a relevance to the real instrument. In contrast, more abstract sound synthesis methods such as FM-synthesis, wave shaping, and others aim to synthesize sound with any method, without any restrictions to the methodology behind the algorithm [99]. These methods also produce synthetic sound, but do not have the controllability of model-based algorithms. However, while model-based algorithms are superior in control issues the sound quality at the moment is still better in sample-based synthesizers.

Within the field of physical modeling of musical instruments lie two schools, reflecting the outlook on computational complexity. One extreme concentrates on being completely physical with the burden of heavy computation and being far from the possibility of practical real-time implementation. The other end has the real-time implementation as an important objective and concentrates on it and usually cuts corners on being completely physical. Both have the goal of producing convincing and natural sounding synthesis. The research of this thesis falls closer to the computationally efficient school of thought.

There are at the moment six categories in the way to do physical modeling or model-based synthesis [100, 9]: source-filter modeling, finite-difference (FD) methods , mass-spring networks, wave digital filters, modal synthesis, and digital waveguides. Next a nutshell overview of the five first methods is given and the last is discussed in Sec. 3.2.

In a source-filter model the source signal is filtered with a time-varying filter. Speech and singing models are examples that contain a physical interpretation of this method [101]. Finite-difference schemes solve partial differential equations, such as the wave equation, numerically. As mentioned, Hiller and Ruiz where pioneers in this field [21, 22]. This line of research has continued [102, 103, 104]. The advantages are that by changing parameters of the model, the properties of the real system can be investigated as the model directly uses physical variables as parameters. However, FD methods are computationally very expensive. The computational cost of mass-spring systems are at the same level as FD methods and combine finite masses to each other through springs and dampers [23, 24]. Even with a high computational cost, the first real-time sound synthesis system (CORDIS) was created with powerful computers and was based on mass-spring systems [23]. Wave digital filters are based on the conversion of analog electronic circuits into digital filters [105]. The thesis by Bilbao studied the application of wave digital filters in the context of modeling of musical instruments [27]. Wave digital filters can also be used for modeling the excitation mechanism of an instrument [106]. Modal synthesis models a system through its modes of vibration that each have a resonance frequency, damping factor, and physical shape described on a discrete grid. Also, modal synthesis can be applied for sound synthesis [107, 108]. The idea of modal synthesis was extended to synthesize percussive sounds by Cook [109]. The functional transformation method is a novel technique related to modal synthesis [110]. It ends up describing a system as a set of vibrational modes after applying two separate integral transformations on the differential equations describing the system.

## 3.2 Digital waveguide models for plucked strings

Digital waveguides (DWG) are a computationally efficient way to model acoustic vibrations in the digital domain [1]. Actually, a proper DWG implements directly the discretized solution of the wave equation Eq. 2.1, namely the traveling wave decomposition of Eq. 2.3. The computational efficiency is based on lumping the distributed delay and loss elements [1]. The DWG theory formulated by Smith [1] can be seen as a generalization of the Karplus-Strong algorithm [111, 112, 4]. For a review see, e.g., [1, 2].

Typically, the DWG models the resonating part of the instrument, such as the string [113, 114, 115, 116] or the tube [117, 30]. The modeling scheme can also be extended to two and three dimensions and in this way be able to model drums and other multidimensional systems [118, 119, 120, 121]. Banded waveguides are another recent extension to the DWG family where the feedback loop(s) contain a cascade of a bandpass filter and a delay line [122].

Based on the LTI assumption of the body of a stringed instrument the so called commuted waveguide synthesis (CWS) method was developed [123, 124]. Here commuting means that the order of the process is changed, i.e., the effect of the body is moved to the beginning (compare with Fig. 2.1). In practice, the crux of the matter is to remove the string resonances from a recorded tone. This way an excitation signal is created which is inserted into the DWG that models the string. Furthermore, the excitation signal contains the interactions occurring during the attack, the effect of the body and radiation, and the nonlinear string behavior not removed during the inverse-filtering process. Hence, the body is inserted into the string, which is the complete opposite of what occurs in the acoustic instrument. This method works especially well for stringed instruments [26, 31].

Figure 3.1 shows the block diagram for a single-delay loop (SDL) digital waveguide filter. It contains the traditional SDL components $H_{\mathrm{LF}}(z)$, $F(z)$, and $A_{\mathrm{d}}$ [4] that have the following purpose. Frequency dependent losses in the string are modeled with a simple loop filter [47]

$$H_{\mathrm{LF}}(z) = g\frac{1+a}{1+az^{-1}}, \tag{3.1}$$

where $g$ is the gain term and $a$ controls the frequency-dependent decay characteristics. The fundamental frequency of the string model is tuned with a fractional delay filter $F(z)$ [125], such as Lagrange interpolator [125]. Inharmonicity and dispersion in DWGs is often modeled with allpass filters $A_{\mathrm{d}}$ [112, 126, 127, 128]. The transfer function of the SDL-DWG is

$$S(z) = \frac{1}{1 - z^{-L_1}F(z)A_{\mathrm{d}}(z)H_{\mathrm{LF}}(z)}, \tag{3.2}$$

where $z^{-L_1}$ is the integer part of the delay line. A complete commuted DWG synthesizer consists of an excitation database, string model, and tonal control units [129], see Fig. 3.2. Alterations and additions are usually added for accurate instrument specific modeling, such as special guitar sounds difficult to model [129] or knocks caused by key release in the clavichord [130].

The effect of the body can be modeled with carefully designed digital filters or reverb algorithms [131, 132, 133, 134, 135, 136, 137, 138]. Body modeling issues and synthesis methods are discussed in [P4], [P5], and [P6]. Model-calibration and extensions to this string model are discussed further in the next section (Sec. 3.3).

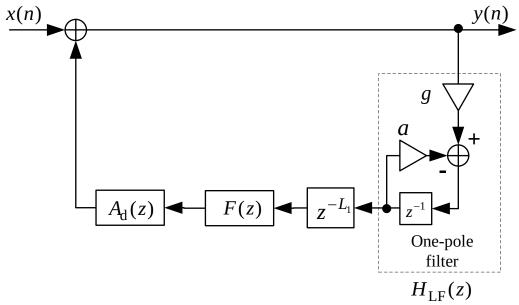


Figure 3.1: Block diagram of a simple one-polarization DWG string model, after [4].

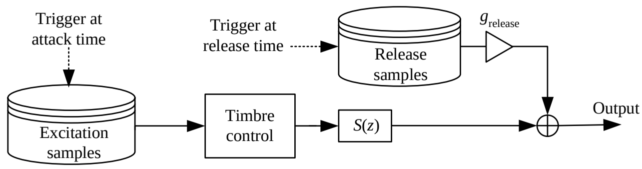


Figure 3.2: Block diagram of a model-based synthesizer with a tone corrector, a string model, and a release sample database, based on [129].

## 3.3 Parameters for digital waveguide models

Parameters for a sound synthesis algorithm can be derived from the physical measurements of the system [113], typically these being a combination of acoustical measurements and recordings of isolated tones [31].

Usually the starting point in automatic parameter estimation is to locate the start of the event, i.e., define the onset of a tone. A simple derivative based method has been used in this work in the context of model estimation. More sophisticated methods do exist [139, 140], and in [P2]-[P3] an onset detection algorithm is proposed which operates in two steps and uses a pitch-synchronized analysis window. After the onset detection the next step in parameter estimation is to estimate the fundamental frequency $f_0$ of the tone. An autocorrelation based technique, the YIN method, was used in this work [141]. Algorithms for estimating the $f_0$s of polyphonic signals are also available [142] and can be used, for example to analyze chords.

Parameters for controlling the decay characteristics of the string model in Fig. 3.1

can be derived analytically from the physical model [143]. Recorded tones can be viewed by the short-time Fourier transform (STFT) analysis to obtain the same information [46, 47]. Design methods for the one-pole loop filter $H_{\mathrm{LF}}(z)$ have been addressed in [46, 47, 137]. Design methods for higher order infinite impulse response (IIR) filters are discussed in [144, 145]. Recently, a higher order finite impulse response (FIR) design method has also been proposed [146]. [P4] proposes the ripple filter extension and [P5] discusses the calibration of the ripple filter for flageolet tone synthesis. Other ways to extract model parameters include parametric techniques [147, 148] and nonlinear optimization methods [149, 150].

The model in Fig. 3.1 is a simplification of what happens in a real string. To be able to model other phenomena than just the frequency dependent exponential decay of partials the model has to be made more complex. In a way, the addition of $A_{\mathrm{d}}$ is such a step. There are methods to model dispersion in stiff strings [112, 126, 127, 151, 128, 152]. As mentioned above, strings exhibit two-stage decay and amplitude beating of decaying partials, see, for example [153, 137] for methods to estimate and model these phenomena. Differences in horizontal and vertical vibrations can be estimated and synthesized with the methods discussed in [4, 137]. Sympathetic coupling between strings is addressed in [4, 154]. In [P4] a simple lumped and unidirectional model for modeling sympathetic vibrations of the highest octave in a harpsichord is proposed.

Nonlinearities in physical models was first addressed in [155]. This was a finite-difference time-domain (FDTD) method, whereas an early nonlinear DWG model was presented in [93] for the kantele. After this, the issue of tension modulation has been investigated further [156, 80, 75, 157]. Nonlinear synthesis methods for bowed stringed instruments and fret-string interactions or nonlinear boundary conditions have also been discussed [158, 159, 160, 161]. Nonlinear models relevant and interesting with regard to this thesis are the ones that consider the synthesis of phantom partials. Many synthesis models for this phenomenon have been proposed recently [162, 163, 16]. Another computationally efficient solution for this purpose is proposed in [P6]. Calibration of DWG synthesis models is addressed and novel synthesis methods are proposed in [P4], [P5], and [P6].

Perception and, more importantly, inaudibility of different features in instrument tones rises to a momentous topic when considering sound synthesis and its computational burden. It is an interesting and useful topic that has been addressed to some extent, see for example [33]. Moreover, this thesis has used the results reported in [33] to make decisions if a feature, such as the initial pitch glide or inharmonicity, should be synthesized or not (see [P4]- [P6]). Additionally, based on the results obtained in [P5], it can be said that harpsichords exhibit observable dynamics, hence this characteristics should be included in high quality synthesis of the instrument.

# Chapter 4

# Summary of articles and main results of this thesis

## 4.1 Publication [P1]

This paper discusses analysis and measurement results of a modified kantele, designed to have an increased loudness. New design rules to make the modified kantele louder are also proposed. The conducted measurements confirm and support the proposed design rules. The design rules suggest (1) to increase the tension of the string, (2) to increase the radiation surface, and (3) to isolate the top plate from the sound-box with an air gap. To some extent rules (1) and (2) are straightforward and familiar to most musical acousticians. In contrast, rule (3) is more evolved and unique since it enables a freely vibrating top plate. To confirm the design rules the traditional design is compared with the new one through analytical treatments and acoustical measurements. Two listening tests were also conducted and the results of these tests support the assumption of an increase in loudness for the new kantele design. More specifically, on the average, loud plucks of the modified design are perceived as 3 dB louder than in the traditional design. Furthermore, on certain strings the loud plucks are perceived as 6 dB louder. It was also found out that the initial pitch glide does not affect the perception of loudness. The proposed design ideas can also be applied to other string instruments.

**Main results:**

- New design of the kantele proven to be louder
- Design rules for making a plucked stringed instrument louder
- Initial pitch drift does not affect the perception of loudness

## 4.2 Publication [P2]

This is the first publication of two that together propose a new method for estimating the plucking point of guitar tones. Previous research concentrated on looking at the phenomenon in the frequency domain. This algorithm relies on investigating the time lag between two consecutive pulses arriving at the bridge of the guitar. In practice, the minimum of the autocorrelation function for one period of the input signal is used. The minimum of the autocorrelation function gives the time lag that can be converted into the plucking point, i.e., the distance from the bridge where the string is plucked. The signal is detected with an under-saddle pickup attached to the bridge. The results obtained with the algorithm are good, so that the error remains smaller than one centimeter, except for a few outliers.

**Main results:**

- Automatic plucking-point estimation algorithm
- New view point to the plucking event and its effect

## 4.3 Publication [P3]

This paper continues the work proposed in [P2] by improving the plucking-point estimation algorithm, providing a real-time implementation of it, and testing the real implementation usage. The improvements were made to the onset detection part of the algorithm by dividing it into two parts and by doing pitch-synchronous analysis. First, the rough onset detection signals a plucking event. Then, the exact detection block looks for the exact moment of the pluck. This is analyzed pitch-synchronously with a window which is half of the wavelength of the fundamental frequency of the signal. This improves the robustness to extract the correct first period of the signal. The real-time implementation is done as an external in the pure data (PD) environment. The testing of the algorithm was executed with the real-time implementation and with a database created for the purpose. The testing has been done for separated signals for reasons of consistency, but the onset detection algorithm is able to detect a new onset also during continuous playing. In addition, the used under-saddle pickup receives the signal of all the strings, hence, for a more practical implementation of the system a six-channel microphone would be needed, one channel for each string. This way all the strings are managed separately and the algorithm is able to process normal playing. The proposed algorithm can be used to control, e.g., audio effects and a sound synthesizer.

**Main results:**

- Improved version of the automatic plucking-point estimation algorithm proposed in [P2] including real-time testing and implementation
- Onset algorithm proposed for plucked tones with pitch-synchronous analysis
- A manner to do real-time control

## 4.4 Publication [P4]

A physically inspired model-based sound synthesis method for the harpsichord is proposed in this article. The synthesis method is computationally efficient and applies the theory of digital waveguide modeling. A modified version of the commuted waveguide synthesis principle is used since the soundboard response is modeled with a separate filter. Also, the ripple filter is introduced which is an alteration to the loss filter of the string model. The ripple filter allows more flexible control over the decay rates of partials than is possible with a one-pole filter, which is the usual choice for the loss filter. A common excitation is inserted into the string model that has a second-order resonator in parallel. The second-order resonator, previously proposed for this purpose, simulates the beating effect appearing in harpsichord tones. The output of the string model is directed to the soundboard model, which simulates the response of the soundboard, the undamped strings, and the ringing of the short parts of the strings behind the bridge. The soundboard response is long ($> 4.5$ s) and modeling it with a filter gives a more natural response than a triggered sample.

**Main results:**

- Efficient parametric model-based sound synthesis method for the harpsichord
- Analysis of harpsichord tones and timbre
- Digital body model filter for soundboard response, the undamped strings, and the ringing of the short parts of the strings behind the bridge

## 4.5 Publication [P5]

The investigations around the harpsichord are continued in this paper by addressing the issue of playing dynamics in harpsichords and by proposing a synthesis model for the phenomenon. In spite of the common assumption, it is shown that the harpsichord contains a limited amount of dynamics and some timbral changes occur when the tangent is pressed down with different speeds. The signal analysis made on recorded harpsichord tones revealed differences in the levels of string harmonics, so that stronger playing forces produced higher levels. The differences for isolated harmonics were as high as 5 dB for some low tones. For higher tones the level differences were smaller, about 1-3 dB. Based on the conducted listening test, it can be said that during each dynamic step (from piano-pianissimo ($pp$) to mezzo-forte ($mf$) and from mezzo-forte to forte-fortissimo ($ff$) the loudness of the instrument increases about 1 dB. In addition to the changes in the level of harmonics, their relative levels differ according to the used dynamic level. Furthermore, $mf$ and $ff$ level tones exhibit soundboard resonances, so that the $ff$ tones have the most prominent soundboard resonances. A general framework for building a model-based synthesis algorithm for the dynamic behavior of the harpsichord is also proposed. Based on this, a digital waveguide model is proposed with a dynamic gain and timbre control, and a dynamic soundboard filter. The dynamic soundboard filter is connected in parallel with the string model to simulate the soundboard knock.

**Main results:**

- Harpsichord observed to contain dynamics and changes in timbre occur according to playing level
- Refined version of the synthesis algorithm proposed in [P4]

## 4.6 Publication [P6]

This paper proposes a sound synthesis model for the ancient Chinese plucked string instrument called the guqin. The guqin has seven strings and is fretless, which enables smooth glides from a tone to another. The body of the guqin is 120 cm long and 18 cm wide hollow box made from two pieces of wooden board, and the top board is carved into an arch while the bottom is flat. The strings are pressed with the left hand against the top board, and hence, it is used as the fingerboard. For pressed tones, one end of a vibrating string is terminated either by the nail of the thumb or a fingertip. The right-hand fingers pluck the strings. Guqin playing also incorporates plenty of flageolet tones in its music. Analysis showed that the string vibrations are inharmonic enough to be audible and that the tones terminated with a fingertip decay faster than those terminated with a thumb. Also, guqin tones exhibit phantom partials. The model-based sound synthesis algorithm uses the digital waveguide approach. A body model filter is placed in cascade with the string model so that when the length of the string changes some of the filtering effect of the body is preserved. Flageolet tones are synthesized with the so called ripple filter structure and a systematic calibration method is introduced. The ripple filter is an FIR comb filter used as an extension to the one-pole loss filter in the delay line of the digital waveguide model. The synthesis model takes into account the important characteristics of the instrument and is able to reproduce them. The synthesis model will be used for rule based synthesis of guqin music.

**Main results:**

- Efficient parametric model-based sound synthesis method for the guqin
- Analysis of guqin tones and timbre. First systematic acoustic analysis of the guqin to the author's knowledge
- Digital body model filter to compensate for time-varying string
- Flageolet tone and phantom partial synthesis and calibration

# Chapter 5

# Conclusions and future directions

This thesis explores issues around plucked stringed instruments, namely their acoustical properties and their sound synthesis. Loudness of the instruments is addressed in the case of the kantele and the harpsichord. A new design of the kantele was proven to be louder, based on acoustical measurements and formal listening tests. Also, loudness changes of the harpsichord was studied and, contrary to common belief, it was found that the harpsichord exhibits dynamics, albeit limited. This issue was investigated through signal analysis and formal listening tests.

The topic of timbre is investigated in the case of the guqin, the harpsichord, and the acoustic guitar. Analysis of timbral features of the guqin revealed that for most of the strings the inharmonicity is large enough to be noticeable and the initial pitch glide is at the boundary of auditory threshold. In addition, the termination technique, fingertip or nail, makes a difference in the decay time of a tone. The exquisite playing of the guqin contains a lot of flageolet tones. A computationally efficient model and calibration scheme is proposed for this purpose. The model-based sound synthesis algorithm for the instrument has covered these characteristics and has also a body model. The timbre of the harpsichord was also approached through its inharmonicity and body response behavior. Dynamic playing was found to affect the timbre of the harpsichord so that an increment in the playing level increases the level of body mode radiation and slightly alters the relative spectral content. A DWG based synthesis algorithm is introduced that accommodates the important features of the instrument. For the acoustic guitar, the plucking point event was addressed, which has a profound effect on the timbre of a tone. As a result of looking at the process from a new point of view, an automatic plucking point estimation algorithm was designed and proposed.

As for future directions of research related to the field of this thesis the themes of sophisticated control schemes, objective sound quality measurements of synthesis, and virtual instrument building can be raised. Naturally, previously not modeled instruments can, will, and should be tackled with the existing methods, and consequently stretch the known algorithms. This would also include improving of already modeled instruments. At the current state, many sound synthesis schemes are able to reach a relatively high level of sound quality. In other words, a synthesized tone is not necessarily distinguished from a recorded one. One of the future steps required is to obtain sophisticated, user friendly, control schemes that make the music played with a synthesis model sound natural. This should occur both in the algorithm and interface domain. That is to say, a gap remains between the rigid tones/music produced with a computer and a human playing an acoustic instrument. Extensive methods for objective sound quality measurement of musical instrument synthesis algorithms, which account for auditory perception (and human taste), are still missing at large. Sound synthesis algorithms have so far been assessed through basic signal analysis schemes, which is well validated. However, formal listening tests, methods, and research around this topic are juvenile or completely missing, whereas research around quality assessment of, e.g., speech codecs is far more mature. Beyond this, a future dream of probably all musical acousticians is to be able to accomplish instrument design in the digital domain. This would mean that one could build and reshape an instrument on the computer and listen to how the changes affect the final instrument before a single finger has even touched wood.

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