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Jan-Markus Holm

Virtual Violin in the Digital Domain

Physical Modeling and Model-based Sound Synthesis of Violin and its Interactive Application in Virtual Environment

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UNIVERSITY OF JYVÄSKYLÄ

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ABSTRACT

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This thesis discusses techniques for physical modeling and model-based sound synthesis of violin, and human-computer interaction in virtual domain. The violin model is considered by means of controlling a virtual violin.

The work can be divided into three parts corresponding to 1) computational analysis and synthesis of audio and music, 2) physical modeling and model-based synthesis of musical instruments, especially bowed strings, and 3) virtual techniques of representing, expressing, and controlling of violin. The emphasis of the work is on physical and physics-based modeling, musical signal analysis and synthesis, and expressiveness and virtual reality techniques in controlling a virtual violin.

The first part of the thesis mainly concentrates on computational audio and music analysis as well as synthesis. It also presents general methods of analysis of harmonic signals using different techniques.

The second part of the thesis considers linear and particularly nonlinear discrete-time modeling of bowed string instruments, especially the violin. The finite width bowing that is present in every excitation phenomenon of a real string has been integrated into the model. Its simulation has been developed into a computationally feasible structure and it has been demonstrated that the models produce desired effects on the synthetic tones in a desired manner. The controllability over the model is considered, and some general outlines on expressiveness of a violin model are carried out.

The third part of the thesis discusses virtual reality techniques in application to human-computer interaction of musical instrument models. An overview on the virtual musical instruments and their control is given, after which the focus is placed on gestural and haptic control of violins. Interaction with the violin model in application to a virtual environment is also discussed.

Keywords: violin, physical modeling, digital signal processing, musical acoustics, nonlinear systems, bowed string instruments, signal analysis, sound synthesis, virtual environment, human-computer interaction.

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PREFACE

This work has been carried out in the Department of Music, University of Jyväskylä, Finland, during 1999-2003. The research has been done in association of Pythagoras Graduate School of Sound And Music Research. In this preface I want to indicate and express my motivation and feelings during the time of giving birth to this thesis.

First of all, I want to thank my primary supervisor, the head and the founder of the graduate school, prof. Jukka Louhivuori, for giving me a hint and throwing me a carrot to apply for a position. The times since the start of the graduate school have been very fruitful; I have been able to be involved in many interesting projects, and a lot of wisdom has been gained during my stay in the Department of Music.

The most crucial person affecting the situation of to date, prof. Petri Toiviainen, deserves my most sincere gratitude for all the guidance and support during the research. Without it, I wouldn’t be at this point today.

I want to thank my graduate school soulmates, Tuomas Eerola and Francis J. Kiernan for sharing research activities, and both good and bad feelings during different organization tasks for the school. I also want to thank all the other scholars, Hanna, Riitta, and Tuire for bringing together a great heterogeneous group of scientists. All the great colleagues at the Department of Music deserve my sincere thanks for being supportive against ‘the stranger’ coming from the technical sciences. The thesis might not be in your hands to read, if prof. Vesa Välimäki from Helsinki University of Technology wasn’t there. It can be said that Vesa gave me the keys for the whole branch of science and kicked off my work. He gave the essential expertise, suggestions, and guidance in the first place. The first paper was co-authored with Vesa.

Furthermore, I want to thank our chamber orchestra Melomania for keeping me on rockin’ by providing an active workbench for making real music with real instruments. Also, I want to thank all my friends, the past and the present, all around the country, especially the old Rautalampi pals for giving nice and relaxing party times every once in a while during this period of research.

To the special people behind, beside, and around me, my mom Eva, dad Alpo, my sister Marjaana and her cute daughter Wilma, as well as my mother- and father-in-law, I wish to express my sincere gratitude for being there, patient and understanding, without questioning why I’m studying for so long not doing ‘real work’. You are the important people in my life.

I dedicate this thesis to the ladies of my life: to my beautiful beloved wife Pauliina and to my lovely little daughter Eveliina. I also dedicate this thesis to my brave little son Henri.

Jan-Markus Holm, Jyväskylä, Finland, 20th April 2004.

LIST OF ABBREVIATIONS

BIFS	BInary Format for Scene description
DWF	Digital Waveguide Filter
DWG	Digital Waveguide
FD	Fractional Delay
FDTD	Finite Difference Time-domain
FEM	Finite Element Method
FFT	Fast Fourier Transform
FM	Frequency Modulation
FOF	Fonction D'onde Formantique
FSR	Force Sensing Resistor
FVEP	Fractional Variable Excitation Point
HMM	Hidden Markov Modeling
KS	Karplus-Strong algorithm
LPC	Linear Predictive Coding
MIDI	Musical Instrument Digital Interface
MLS	Maximum Length Sequence
MPEG	Moving Picture Expert Group
MQ	McAulay-Quatieri algorithm
PDE	Partial Difference Equations
SDL	Single Delay Loop
TMS	Transient Modeling Synthesis
VOSIM	Voice Simulation
WDF	Wave Digital Filter
WT	Wavelet Transform

LIST OF PUBLICATIONS

This thesis summarizes the following articles and publications, referred to as (I) – (VI):

I Holm J.-M. and Välimäki V. 2000. Modeling and modification of violin body modes for sound synthesis. Proceedings of the European Signal Processing Conference, Eusipco2000, vol. 4, Tampere, Finland, 2229-2232.

II Holm J.-M. 2001. Towards complete physical modeling synthesis of performance characteristics in the violin. Proceedings of the VIII Brazilian Symposium on Computers and Music, Fortaleza, Brazil.

III Holm J.-M. 2002. An FD finite width excitation modeling approach in violin sound synthesis. Proceedings of the Audio Engineering Society 22nd International Conference on Virtual, Synthetic and Entertainment Audio (AES22), Espoo, Finland, 14.-17.6. Audio Engineering Society, 256-261.

IV Holm, J.-M. 2002. Estimation of frequency dependent damping and implementation for physical modeling sound synthesis of violin. Proceedings of the International Computer Music Conference, 17.-21.9. Gothenburg, Sweden, 280-282.

V Holm, J.-M. 2003. Real-time tracker controlled virtual violin. Computer Music Journal, The MIT Press. Submitted in April 2004.

VI Holm, J.-M. and Toiviainen, P. 2003. Modeling the finite width excitation in physical modeling sound synthesis of bowed strings. Journal of New Music Research. In Press.

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> "I prostrate myself in front of the patriarch of violin makers. … If in his century a competition had been staged in which all the great violin makers had been judged by their best works, the five Amatis would have obtained honourable mention, Stainer would have been runner-up, but without hesitation and unanimously, Stradivari would have been awarded the prize. … The first six are simply admirable, each one in a particular aspect of the art, while the last is perfection itself."

(A history of violin making written by the Abbé Sibire (1757–1827), *La Chélonomie, ou Le parfait luthier* (Paris, 1806), a paean to Stradivari.)

> "I shall confine myself hereafter to a daily occurrence …. It is a kind of restoration (loosely called) which is purely accessory and yet at the same time crucial. This is a process which does not imply the slightest deterioration and yet which virtually every old violin, no matter how well preserved it is in other ways, could not avoid: REBARRING. The revolution which music has experienced needs to be replicated in instrument making; when the first has set the style, the other must follow. … Formerly it was the fashion to have necks well elevated, bridges and fingerboards extremely low, fine strings, and a moderate tone. Then the bass bar, that necessary evil in the instrument, could be short and thin because it was sufficient for it to have enough strength to sustain the weight of five to six pounds which the strings exerted on it. But since then music, in becoming perfect, has placed a demand on violin making. The tilting back of the neck, the raising of the bridge, of the fingerboard, and the amplification in sound, necessitate increasing by a full third the resistant force. Repairers have only one choice: strengthening the old bar, or replacing it with a new one."

(The Abbé Sibire in 1806 in *La Chélonomie*. The description of structural changes as a response to changes in musical expression.)

1 INTRODUCTION

The violin is undoubtedly one of the most extraordinary and complicated acoustical devices ever created by and for human. This amazing product of the developments of the Baroque era has been the subject of scientific research since the early 19th century. Even now, however, the complex acoustical interrelationships between the parts of the instrument are incompletely understood. With each technological advance more has been discovered about the violin works, although the total context of a musical instrument, including the player, listener tastes, hall acoustics, as well as the challenges of the new technology bringing along new innovations is researched more intensively. After this all we will understand the subtleties involved in this seemingly simple, yet highly sophisticated wooden box with strings attached, and perhaps even continue the creation of new instruments of same kind by association with the modern technologies.

In the last decade, new techniques for digital signal processing in musical modeling and sound generation have rapidly gained popularity. These methods can be generically referred to as *physically-based*, since the synthesis algorithms are designed by modeling the physical mechanisms that underlie sound production. At the same time, high quality digital audio converters have become standard hardware on personal computers, and the available computational power permits real-time implementation of these synthesis algorithms. Research in audio and acoustics has shown that such models can provide convincing results and physically consistent control over the synthesis algorithms.

An important application of physically-based sound models is used in human-computer interaction, where tactile or force information can be exchanged through suitable sensors and effectors. In this case, auditory information can greatly augment the sense of presence of the user and the realism of the interaction. In this respect, physically-based synthesis is advantageous over other techniques, such as sampling, for two main reasons. Firstly, the physical description of the sound algorithms allows the user to interact with the sound objects. For example, in a physical model of contact the friction sound of the user’s hand on a surface changes with the pressure exerted by the user. Likewise, the sound produced by a struck object varies with the impact velocity. Secondly, physically-based models for audio and graphics can be, in principle, easily synchronized. This way, a high degree of perceptual coherence of acoustic and visual events can be achieved. The efforts toward the development of models for joint audio-visual synthesis are rewarded in terms of simplification in the design of multimedia systems.

Virtual musical instruments can be regarded as a particular case of human-computer interaction. When using an acoustical or electro-acoustical instrument, the player interacts with it in a complex way and exerts control by exchanging gestural, tactile, and force information. Techniques used in commercial synthesizers are mostly based on wavetable methods (i.e., recorded and post-processed sounds, see section 2.3) that allow little sound manipulation. Consequently, the only interface that has been widely used so far in commercial electronic instruments is the piano keyboard. Again, this interface provides the user only with very little control over the synthesized sounds. In this respect, physical models of musical instruments greatly improve the possibilities of interacting with the virtual instrument.

As an example, sound production in a physical model of the violin is controlled by parameters such as bow velocity and pressure. The design of accurate and efficient physical models of musical instruments encourages the development of more sophisticated interfaces that in turn give the user access to a large control space.

One advantage of the physical modeling approach with respect to other techniques (such as concatenative synthesis or LPC-based analysis/synthesis method, see sections 2.2 and 2.3) is that more realistic signals can be obtained. Moreover, the models can, in principle, be controlled using physical parameters, such as blowing pressure in clarinet playing, and tension of the vocal folds in singing. However, it must be stressed that the problem of control is still an open one, since finding direct mappings between the physical parameters, expressive gestures and musical cognition, and perceptual dimensions, such as loudness, pitch, and register, is not a trivial task.

The first subsections in each chapter review the existing literature and discuss the general techniques in use. The rest of the sections contain deeper discussion concerning the actual research of the author in the scope of corresponding studies.

Chapter two describes a framework of physical modeling and model-based synthesis of musical instruments. The general analysis and synthesis methods are shortly discussed. The framework also clarifies the overall field of research and the relations between the studies reported in publications. The analysis part of the framework presents the author's contribution to the analysis of violin providing a basis for model-based synthesis (IV).

Chapter three summarizes the author’s contribution to the physical modeling and model-based sound synthesis in application to the violin. More precisely, the chapter discusses model-based synthesis including linear and non-linear models and their applicability for the violin. The chapter summarizes publications (I-IV, VI) describing the investigations on different parts of the system of the violin, for instance, the finite width bowing and body modeling. Furthermore, expressiveness and control over the model, discussed in publication (II) is considered. In the co-authored papers (I) and (VI) the present author has been responsible of the measurements, creation, and implementation of the models, and carrying out the simulations, while the text itself has been co-authored.

Chapter four presents the violin model in application to interactive virtual technologies. The physical model of the violin discussed in the previous chapters is applied in gestural control, and the paper (V), as well as the contribution of the author is summarized.

Finally, chapter five presents conclusions and directions for future research.

2 PHYSICAL MODELING AND MODEL-BASED SYNTHESIS OF MUSICAL INSTRUMENTS

Physical modeling of musical instruments is an exciting paradigm in digital sound synthesis. The basic idea is to imitate the sound production mechanism of an acoustic musical instrument using a computer program. The sound produced by such a model will automatically resemble that of the real instrument. One requirement is that the model has been devised in a proper way.

This chapter presents an overview of the history and present techniques of sound analysis and synthesis, such as physical modeling. It appears that the many seemingly very different modeling methods try to achieve the same result: to simulate the solutions of the wave equation in a simplified manner.

In this thesis the concentration is on the digital waveguide modeling (DWG) technique (see section 2.3), which has gained much popularity among both researchers and engineers in the music technology industry. The benefits and drawbacks of different technologies are considered and concurrent research topics are discussed. The physical modeling approach offers many new applications, especially in the fields of multimedia and virtual reality.

The motivation in developing physics-based models is generally twofold. The first is that of science, in which models are used to gain understanding of physical phenomena that take place in musical instruments. In spite of all the research, we are still lacking details in our understanding of musical instruments (e.g. Weinreich 1993). Hopefully, physical modeling will reveal more of this fascinating area where art, craftsmanship, and science meet. The other part is the production of synthesized sound. From the days of first physics-based models researchers and engineers have utilized them for sound synthesis purposes. Presently, many commercial synthesizers simulate mechanisms of actual musical instruments.

The basic idea of the modeling is that when the vibrating structure is simulated in exactly the right way, the sound produced by that model is identical with the sound of the corresponding physical object. The physical structure of a musical instrument is being modeled with mathematical and physical formulae that are realized as algorithms on a computer.

The general difference between synthesis using physical modeling and other synthesis techniques is that the former tries to imitate the properties of the sound source, while the latter focus on the properties of the sound signal heard by the listener (waveform, spectrum, etc.).

The main advantages of physical modeling synthesis are that the parameters of the technique are physically meaningful, such as the bowing velocity in bowed string instruments, and important parts of the evolution of single tones, for example the attack and decay, are generated automatically in a correct way.

Physical models are naturally developed in the continuous-time domain and are described through sets of ordinary and partial differential equations. In a subsequent stage, these equations have to be discretized. Accurate techniques are needed to minimize the numerical error introduced in the discretization step, to guarantee stability of the numerical algorithms, and to preserve as closely as possible the behavior of the continuous systems. At the same time, the numerical techniques have to produce efficient algorithms that can be implemented in real time.

These two demands of accuracy and efficiency often require making trade-offs. As an example, implicit and iterative discretization methods guarantee sufficient accuracy but affect the efficiency of the resulting algorithms. Likewise, low sampling rates are preferable for efficient implementations, but the accuracy deteriorates.

The first commercial products based on a physical model appear to be the Yamaha VL-1 introduced in 1994, after which Korg introduced a related product, the Wavedrum in 1995, and then Roland the VG-8 guitar processor. Since then, simplified software-only implementations of musical instruments have appeared. Of these, only the VL-1 is a purely physical model. The others use sound processing algorithms that are motivated by the principles of physical modeling.

In addition to physically related principles it is possible to combine physical models with other synthesis and signal processing methods to realize hybrid modeling techniques. An overview of some recent results in model-based sound synthesis and related signal processing techniques is given by Karjalainen (1999).

2.1 Analysis methods

In this section, an overview on the sound analysis techniques is given. The emphasis is on discussion mainly from the point of view of physical modeling and model-based sound synthesis, since the signal analysis techniques are applied in physical model synthesis parameter estimation. In the sense of measured instrument sounds, the section concentrates only on monophonic signals. The techniques in application to automatic transcription, musical information retrieval, as well as tempo and beat tracking are out of the scope of this thesis. It is crucial, however, to analyze and investigate the expressive parameters in musical instrument modeling. Instead of discussing the topic in this section, it is separately covered in Section 3.3.

Already in the 19th century, musical instrument tones were divided into their Fourier series. Early techniques for the time-varying analysis of the additive parameters are presented by Matthews *et al.* (1961) in application to speech, and by Freedman (1967) in application to musical instruments. Spectrum estimation methods have been discussed by Robinson (1982). Other techniques for the analysis of musical signals are the proven heterodyne filtering (Grey *et al.* 1977), the wavelet analysis (Kronland-Martinet *et al.* 1991), the atomic decomposition (Goodwin *et al.* 1999), and the cepstrum technique (Cappé *et al.* 1995). The FFT-based analysis has been suggested, for instance, by McAulay *et al.* (1986), and the linear time-frequency analysis by Guillemain *et al.* (1996). Generally, many of the synthesis methods (see Section 2.3) are also used for analysis purposes.

In the following sub-sections, some analysis methods are further discussed and within some of the methods their contribution to this thesis is explained.

Sinusoidal Analysis/Synthesis

Sinusoidal modeling was originally a set of techniques in which a sound signal is represented as a set of sinusoids that are parameterized by amplitude, frequency, and phase trajectories (McAulay and Quatieri 1986, Smith and Serra 1987), which was later accompanied with models for noise (Serra and Smith 1990) and for transients and noise (e.g. Verma *et al.* 1997). Iterative sinusoidal analysis algorithms have been presented by e.g. George and Smith (1997) and schemes using different time resolutions in different frequency bands by e.g. Goodwin (1997). Improvement to the estimation of the sinusoidal components is proposed by e.g. Depalle and Hélie (1997). Also the Hidden Markov Models (HMM) are applied in trajectory estimation by Depalle *et al.* (1993) that facilitates determination of crossing partials.

In general, the peak continuation algorithm of sinusoidal modeling is one of the crucial parts of the analysis system. Similar approach for STFT parameter estimation using HMMs is described, for instance, by Streit and Barrett (1990).

Sinusoidal methods have originally been applied in the coding of speech, although later sinusoidal modeling methods have been applied to speech synthesis (Macon and Clements 1996) and singing voice synthesis (Macon *et al.* 1997). Maher and Beauchamp in 1990 improved the technique for analysis of vibrato (Quatieri and McAulay 1998). For musical signals, sinusoidal modeling analysis and synthesis techniques have been used, for instance, by Smith and Serra (1987), and Serra and Smith (1990), and additive deterministic plus stochastic sine-wave representation by Serra and Smith (1990).

Some improvements in frequency resolution have been searched using Prony's method (Laroche 1989) where the technique was applied to a variety of musical sounds, such as marimba, bell, and vibraphone. Variations of the method have been investigated by Laroche (1994). The constant Q analysis/synthesis exploits auditory spectral masking attempting to reduce the number of sine-waves (Anderson 1996).

Sinusoidal model has been suggested also in analysis of least-damped modes of string instrument body impulse response (Karjalainen and Smith 1996), as often used to determine the string loop filter in waveguide models (see the following section). The methodology of ARMA modeling by pole-zero filters for measured impulse responses has been investigated by Karjalainen et al (2002). In addition to an overview of the standard AR and ARMA techniques, a spectral zooming technique has been proposed, which is useful for resolving very closely positioned modes and high-density modal clusters. Application case related to The estimation of parameters for musical instrument modeling has been studied. A detailed review of the sinusoidal techniques is given by Quatieri and McAulay (1998).

SMS analysis

Spectral Modeling Synthesis (SMS) technique is used, not only as a synthesis (see also section 2.3) but also as an analysis tool. In this method the signal representation is given by the spectral envelopes of the stochastic[1] component of the input signal. The envelopes are calculated from each *Discrete Fourier Transform*[2] (DFT) frame (see section 2.3). The analysis follows similar tracks to the McAulay-Quatieri (MQ) algorithm (1986). The portion of windowed *Short Time Fourier Transforms* (STFTs, see section 2.3) produces a series of complex spectra from which the magnitude spectra is calculated. Furthermore, the prominent peaks are detected and the peak trajectories are obtained by peak continuation algorithm.

Overlap and add method

A variation of granular synthesis technique (see Section 2.3) consists of analyzing each windowed segment and re-synthesizing each of them with a method called *overlap and add* (OLA). OLA-based methods are efficiently used for synthesis of random component of a signal (Serra 1997).

Methods based on waveform similarity in analysis process are the *Synchronized OverLap-Add* (SOLA) originally proposed by Roucos and Wilgus in 1985 (Laroche 1998) and its variations, mainly in speech processing (Wayman and Wilson 1988, Verhelst and Roelands 1993), and the *Pitch Synchronous OverLap-Add* method where the length of handled signal segments is adjusted

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[1] Deterministic and stochastic components of a signal are discussed in detail by Serra and Smith (1991).

[2] Discrete Fourier transforms (DFT) is usually computed by Fast Fourier Transform (FFT) algorithm (see e.g. Allen 1977).

according to the local value of the pitch given by a preliminary pitch estimation (Laroche 1998).

Wavelet analysis

Wavelet transform (WT) was originally developed for applications in physics and acoustics (Kronland-Martinet and Grossman 1991, Godsill and Rayner 1997, Vetterli 1992). A wavelet is a signal forming a sinusoid with a smooth attack and decay (Roads 1996). In the theory of wavelets, every input signal can be expressed as a sum of wavelets having a precise starting time, duration, frequency, and initial phase. A prototypical wavelet has a Gaussian envelope in music purposes. The wavelet window expands or compresses, *dilates*, according to the frequency being analyzed with a factor of 1/*frequency*.

Linear predictive coding

In *Linear Predictive Coding* (LPC) analysis it is possible to estimate both the parameters of the source and the filter of a given sound. By analyzing brief sequential segments of the sound, time varying parameters can be extracted and used in re-synthesis. For instance in string instrument body modeling (Karjalainen and Smith 1996), the linear prediction is used for converting the least-damped resonating modes into parametric form.

Other analysis methods

Certain other methods are also applied to analyze sound and vibration signals. A few of them are shortly mentioned here. The *Time Domain Finite Difference* (FDTD) techniques (see section 2.3) have been extended to analyze struck strings of piano (Chaigne and Askenfelt 1994), vibrating bars and resonators of a xylophone (Chaigne and Doutaut 1997), and damped impacted plates (Chaigne and Lamberg 2001, Lamberg *et al.* 2001). *Modal analysis* is the process of describing the dynamic properties of an elastic structure in terms of its normal modes of vibration. A general overview on modal analysis of musical instruments is given by Fletcher and Rossing (1998). The methods are discussed in more detail in the contexts of sound synthesis in section 2.3 and of violin parts, such as analysis of violin body, in section 3.1.3.

Sound analysis for model parameter estimation

The fundamental frequency of a musical sound is one of the most important attributes to be analyzed here. Several algorithms for estimation of fundamental frequency have been presented in the last few decades. The fundamental frequency estimation can be done in the time domain (Rabiner *et al.* 1976, Rabiner 1977, Kroon *et al.* 1990), in the cepstrum domain (Noll 1967), or in the frequency domain (Doval *et al.* 1991, Jensen 1999). A proposal for the objective comparison of pitch estimation techniques from a database of a wide range of sounds is given by Freed *et al.* (1997).

A popular approach in the estimation of the fundamental frequency is windowed autocorrelation function (Rabiner 1977). An estimate for the pitch is obtained by searching for the maximum of this function for each time frame. The method has been applied widely in musical instrument sound analysis.

Several methods for tracking partials have been developed, such as local optimized techniques (McAulay *et al.* 1986, Serra *et al.* 1990), or globally optimized techniques using, for instance, Hidden Markov Modeling (HMM) (Depalle *et al.* 1993).

On the analysis of instrument sounds it is very efficient to use time-frequency analysis, thus analyzing the sound spectrum as function of time. This is achieved using the *short-time Fourier transform* (STFT) and tracking the amplitude of each harmonic. The STFT of a signal is a sequence of discrete Fourier transforms (DFT) that is usually computed by Fast Fourier Transform (FFT) algorithm (Allen 1977), and sometimes with the addition of a stochastic component model of additive noise (Serra *et al.* 1990). When the values of the Fourier transforms are expressed in polar coordinates, the STFT is alternately called the phase vocoder (e.g. Laroche 1998). Historically, a running Fourier transform to measure sine-wave amplitude and phase trajectories evolved into signal analysis-synthesis method referred to as the phase vocoder introduced by Flanagan and Golden in 1966 (Quatieri and McAulay 1998). On the other hand, heterodyne filtering has also been used in application to guitar sound partial tracking (Välimäki and Tolonen 1998) taking advance of deterministic and residual components (Serra and Smith 1990).

The system identification for a waveguide string instrument model typically consist of 1) estimation of the body model, 2) estimation of the loss filters of the string models, and, in impulse excited instruments, 3) estimation of an excitation sequence (e.g. Karjalainen *et al.* 1993). In the third case, inverse filtering approach may be used.

The system identification and the loop-filter calibration for a recorded sound of vibrating string were studied by Smith (1983) as one of the first attempts in string modeling. The STFT method has been used in musical instrument modeling to estimate, for instance, harmonic damping and furthermore, loop-filter parameters for string model in application to plucked strings (e.g. Karjalainen *et al.* 1993), and especially in case of extended *Karplus-Strong* (K-S) model (Karplus and Strong 1983, Jaffe and Smith 1983, Karjalainen *et al.* 1993). In the estimation process for the loss filter of a string model the decay rates for individual partials are detected, the gains of each slope are computed, and a digital filter matching the amplitude and decay properties is designed (e.g. Karjalainen *et al.* 1993, Tolonen and Välimäki 1997). In these studies, different designs of loop filters have been investigated. Earlier, Smith (1982) proposed loop-filter estimation using deconvolution, though the method results in very noisy estimates. Analysis and estimation methods similar to Karjalainen, Tolonen and Välimäki have been used in this thesis in application to sound analysis for frequency-dependent damping model of violin. The frequency dependent damping estimation, filter design, and implementation for violin physical model are discussed in paper (IV). The synthesis methods mentioned above are described in more detail in section 2.3.

Body modeling for string instrument synthesis has been extensively studied by Karjalainen and Smith (1996). They have listed various methods for body mode parameter estimation, such as amplitude response peak measurement, weighted digital filter design, linear prediction, sinusoidal modeling, and late impulse-response analysis. The violin body analysis and modeling is discussed in sections 3.1.3 and 3.2.2, and contributed in paper (I).

2.2 Synthesis methods

The first experiments in synthesis of sound by computer were begun in 1957 by Max Mathews and his colleagues at Bell Telephone Laboratories (Roads 1980, 1989). The aim was to prove that a computer could synthesize sounds according to any pitch scale or waveform, including time-varying frequency and amplitude envelopes. Since the development of Mathews' first synthetic instrument[3], enormous efforts in the field of music synthesis have been made.

A sound synthesis technique maps time-varying musical control information into sound. Each different synthesis method can be evaluated not only in terms of the class of sounds it is able to produce, but also in terms of the musical control it offers to the musician. However, certain fundamental ideas for sound synthesis are shared by multiple techniques.

From the point of view of signal theory (see e.g. Ifeachor 1993, Orfanidis 1995), a satisfactory numerical representation of sound is obtained by sampling the pressure signal at a sufficiently high rate. Since the human auditory system is limited mainly from 20 Hz to 20 kHz being largely insensitive to frequencies above 20 kHz, any sampling rate higher than 40 kHz provides adequate numerical sound signals. Depending on the application, even lower bandwidths can be used. Speech signals are traditionally coded using low sampling rates, since the focus is often on intelligibility rather than quality. Sampling in time the sound signal does not assume any underlying structure, or process, or generative model, in sound representation. The symbolic description is extremely poor, and as a consequence very little interaction with the sound representations is allowed. Although sophisticated signal processing techniques are available that provide meaningful modifications (e.g. pitch shift, time stretching), sampling is basically a *static*, low-level description of sound.

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[3] In 1970, Mathews pioneered the GROOVE system (Generated Real-time Output Operations on Voltage-controlled Equipment), the first fully developed hybrid system for music synthesis, utilizing a Honeywell DDP-224 computer with a simple cathode ray tube display, disk and tape storage devices. The synthesizer generated sounds via an interface for analog devices and two 12-bit D/A converters. Input devices consisted of a "qwerty" keyboard, a 24-note keyboard, four rotary knobs, and a three dimensional rotary joystick. (e.g. Kahrs 1998)

In the beginning, sound synthesis techniques were divided into two categories according to linearity. An acoustic system is said to be linear if (I) the sum of outputs produced by two different input signals is the same as the output signal produced when the sum of these two signals has been used as an input, and (II) the amplification of the input signal by some factor causes the output to be scaled by the same factor (Ifeachor 1993, Orfanidis 1995). These are known as the principles of superposition and homogeneity, respectively. All the techniques that do not fulfill these principles are called nonlinear. Some basic features of nonlinear systems are that the output signal may contain other frequencies than those present in the input and that the spectral content of the output signal depends on the amplitude of the input signal. The division to linear and nonlinear is no longer valid due to the techniques being developed. Now they are mixtures of various techniques with linear and nonlinear elements appearing as components of the techniques.

Higher-level representations of acoustic signals are necessarily associated with some abstract paradigms that underlie sound production. In trying to categorize the sound synthesis methods, the first distinction can be traced between *signal models* and *source models* (Smith 1991, DePoli 1991). Any algorithm that is based on a description of the signal in the time-domain or in the frequency-domain, and makes no assumptions on the generation mechanisms belongs to the class of signal models. On the other hand, Borin *et al.* (1997) have roughly divided sound synthesis algorithms into two broad classes: classic direct synthesis algorithms, including sampling, additive, granular, subtractive, and nonlinear transformation synthesis, and physical modeling techniques containing methods that model the acoustics of musical instruments (Borin *et al.* 1997).

In his classification of sound synthesis techniques, Smith (1991) suggested the techniques to be put into four categories: (a) Abstract algorithms, (b) Processing of recorded samples, (c) Spectral models, and (d) Physical models. On the other hand, Kahrs (1998) has divided synthesis techniques into other four basic categories: (i) Additive (Linear), (ii) Subtractive, (iii) Nonlinear, and (iv) Physical modeling. Here, the first categorization is considered and divided into two main categories having few subsections including the synthesis methods as follows: *Signal Models* consisting of (a) Abstract (FM, Nonlinear methods), (b) Temporal (Granular, Wavetable), (c) Spectral (Additive, Subtractive, SMS, LPC), and *Source Models* consisting of (d) Physical models (DWG, WDF, Lumped models, Finite differences). Serra (1997) has categorized the musical sound generation models generally into three basic types: instrument models (physical models), spectrum models, and abstract models.

In the following sub-sections an overview on the main classes of digital synthesis techniques is given. In the last part of this section, the physical modeling synthesis is discussed generally. Moreover, being the main simulation method used in this thesis, the physical modeling and model-based sound synthesis techniques in application to violin are discussed in Section 3. As a conclusion to the sound synthesis techniques we will see that many of the principles used in traditional sound synthesis also play an essential role in physical models.

Abstract modeling

In this section the abstract modeling techniques are discussed. The techniques of this class can be described to include methods that produce sound based on a mathematical formula which has no direct relation to real-world sounds or acoustic principles of sound generation, though aiming to reconstruct complex dynamic spectra. This implies that it is difficult to predict what kind of sound is produced by a particular algorithm, or to design an algorithm that results in a particular sound. The nonlinear synthesis techniques are commonly called modulation techniques since they usually employ multiplication of two signals.

The traditional nonlinear techniques, such as *FM synthesis* and *waveshaping* are well-known representatives of this class. Other synthesis techniques in this large group are, for instance, *granular synthesis* (Roads 1996) (see the following section) and *scanned synthesis* (Verplank *et al.* 2000).

Frequency Modulation (FM) was originally described by Chowning (1973), after which it was licensed to Yamaha and used in the legendary DX-7 synthesizer[4]. FM synthesis is really a completely abstract representation, its main control parameters being a carrier and a modulating frequency. FM instruments are made from cascades of FM oscillators where the outputs of several oscillators are mixed together. It is quite well know that controlling an FM algorithm is far from being an intuitive task, and small changes in the parameters can result in large timbral distances in the sounds (Roads 1996).

Waveshaping synthesis, also known as nonlinear distortion, uses a nonlinear shaping function to modify the input signal. In the waveshaping method (Arfib 1979, Le Brun 1979, Roads 1996) a signal, such as a sine wave, is fed into a function that maps the amplitude values in a nonlinear manner. An example of this principle is the clipping caused by overdriving a guitar amplifier. Here the large amplitude values are compressed, while the small values are not much affected (Välimäki and Takala 1996). Beauchamp (1979) applied this method to produce brass instrument sounds from pure sine waves. Later, waveshaping has been categorized as a part of certain physical models.

Sampling

In this section we shortly discuss the methods that are categorized into the group of temporal synthesis methods. *Sampling wavetable synthesis*, also known as *sampling* is one of the commercially most popular music synthesis techniques in use. The basic principle is to record and store large databases of waveforms, after which they are processed and played back.

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[4] The Yamaha DX-7 released in 1983 was the first product to use the FM principle and also the first fully digital synthesizer. It represented a breakthrough for digital sound synthesis.

The technique offers a large variety of possibilities: any sound, acoustic or synthetic, can be recorded digitally, filtered or edited, or combined with other signals, and finally the processed version can be listened to. The processing can be so powerful that the original sampled sound may not even be recognizable, although it still affects the result. This principle of processing recorded sounds is also a useful tool for physical modeling.

Wavetable synthesis is able to provide high sound accuracy, but offers very little flexibility and expressive freedom. Manufacturers have implemented the method largely to commercial products, synthesizers[5]. The technique is very well suited to keyboards that have very few controls, such as note on/off, pitch, and velocity. While in samplers the playable signal is recorded in its whole length, in wavetable synthesizers the stored waveform is repeated in order to have a periodic tone. The original samples are synthesized with some other technique, such as additive or subtractive synthesis, for instance. More detailed overview on the sampling wavetable synthesis is given by Massie (1998).

Karplus and Strong (1983) presented a method for synthesis of plucked string and drum sounds that turned out to provide exceptionally good quality in sound synthesis. The *Karplus-Strong* (K-S) technique may be seen as an extension to the wavetable synthesis. Opposed to the technique of recorded instrument tone periodically repeated, in the K-S method the wavetable is initialized with a sequence of random numbers. By means of digital signal processing, the K-S algorithm basically forms a comb filter. After the first proposal of the method it has been widely studied. A physical modeling interpretation of the K-S algorithm has been presented by Smith (1983) and Jaffe and Smith (1983). Extensions to the K-S method are discussed in section 2.2 from the viewpoint of signal analysis, and in the physical modeling section of this chapter.

Granular synthesis can be seen to belong to this group of synthesis techniques partly due to its representation in both time and frequency scales, although it is usually grouped with abstract synthesis techniques, as well as partly spectral modeling techniques, as suggested by Risset and Wessel (Roads 1996). The granular synthesis can be compared to the dualistic property of light. It builds up acoustic events from thousands of sound *grains*. As constellations of elementary units of energy, with each unit bounded in time and frequency, the representation is a useful way of viewing complex sound phenomena (Roads 1996). Such representations are commonly used in synthesis and signal processing algorithms; thus very similar procedures have different names: *short-time segment* (Schroeder and Atal 1962), *short-time weighting function* (Flanagan 1972), *wavelet* (Kronland-Martinet and Grossman 1991), *formant-wave-function* (Rodet 1984), and *VOSIM pulse* (Kaegi and Tempelaars 1978) can all be described as granular representations of musical signals (Roads 1996).

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[5] Switched wavetables have been used in commercial products, such as PPG, Korg Wavestation, and Ensoniq VFX.

Spectral modeling

The category of spectral modeling techniques includes additive and subtractive synthesis, formerly called also linear synthesis methods, and SMS and LPC (see also section 2.2) synthesis in application to a few new approaches that have been developed so far. In particular, these methods utilize the fact that interesting timbres are not stationary but are characterized by the evolution of their spectral components.

This widely accepted sound synthesis technique, *spectral modeling*, has been qualified very powerful attempting to describe the sound as it is perceived by the ear. Unlike most methods, it allows spectrally-based effects, such as sound morphing to be applied. Closely related to spectral modeling, signal processing technique based on sinusoidal analysis/synthesis (e.g. Quatieri and McAulay 1998) is discussed in the analysis part of the thesis in section 2.2. The origin of sinusoidal modeling that was given in application of speech processing is discussed by Quatieri and McAulay (1986). This analysis-based representation, also known as *McAulay-Quatieri* (MQ) *algorithm*, was shown to be capable of synthesizing also a broader class of sound signals. The general historical backgrounds of the method have been discussed by McAulay and Quatieri (1986).

Additive synthesis is one of the oldest and most investigated synthesis techniques in computer music, and has been extensively described by Moorer (1977). It is a good example of spectral synthesis: it exploits the Fourier theorem[6] to describe a complex sound as a sum of elementary waveforms, such as sinusoids, each controlled in amplitude, frequency, and possibly phase. By means of computed Fourier Transform, the Fast Fourier Transform (FFT), the additive synthesis can be efficiently realized. First, the spectrum of the signal is defined and then converted into a time waveform using the inverse FFT algorithm. Rodet and Depalle have shown the possibility to reduce the amount of data in the spectral representation and yet produce high-quality musical tones via the inverse FFT (Rodet 1996, Välimäki and Takala 1996). Using the Short-Time Fourier Transform (STFT) as the main operational tool, additive synthesis can successfully model quasi-periodic sounds. As such, the additive synthesis is not suitable for synthesizing noise-like signals, such as a trumpet tone containing a great deal of "breath". In spite of that, the noisy component is proposed to be handled stochastically with colored noise.

Serra (Serra 1997, Serra *et al.* 1997) proposed an extended analysis/synthesis technique, named *Spectral Modeling Synthesis* (SMS) that includes a time-varying noise component. The method for deterministic and stochastic components in signal has been introduced by Serra and Smith (1991), discussed by Serra (1994), leading to reconstruction-plus-residual decomposition. The approach, labeled as *sines plus noise model*, is similar to the sinusoidal analysis-synthesis process, where the deterministic part is likened to

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[6] Fourier's theorem states that any periodic waveform can be modeled as a sum of sinusoids at various amplitudes and harmonic frequencies.

the residual of the sum of partials and the stochastic part is similarly likened to the residual of the sinusoidal analysis-synthesis process. That is to say, in the synthesis the harmonics and other prominent frequency peaks of a sound are imitated by sine waves, and the rest of the sound (the residual) is modeled as noise. The model helps considerably in music synthesis, especially since re-synthesis of a noisy musical tone was one of the weakest points of traditional additive synthesis because a vast number of sine waves were required to reproduce broadband noise. In the sines-plus-noise model, the noise component is created by filtering white noise with a time-varying filter. As an extension to SMS the *Transient Modeling Synthesis* (TMS) has been presented by Verma *et al.* (1997). In this method, the residual signal is represented in two parts, transients and steady noisy components. A parametric representation of transient components is provided.

A major drawback of additive synthesis is its enormous number of control parameters: at least one amplitude and one pitch envelope has to be specified for each partial. Moreover, the sound representation has not a strong *semantic* interpretation, since these parameters do not have a high-level meaning. In practice, the applicability of additive synthesis has been pointed out when sounds with specific characteristics are required, as in psychoacoustic experiments (Borin *et al.* 1997), for instance.

Subtractive synthesis refers to linear filtering of an input signal so that some frequency components are attenuated (subtracted) and some emphasized. A better term is *source–filter model* since the basic idea of subtractive synthesis is that sound production is divided into two parts: an excitation signal (source) and a resonator (filter). The excitation is typically a spectrally rich waveform, such as noise or some periodic pulse-form whose spectrum has energy at several frequencies. This is what happens when the vibrations of violin strings are transmitted to the resonant body of the violin, for instance. The filter not only attenuates some of the frequency components of the input signal but may also boost some of them, and can therefore be used to imitate formants, which are resonances that characterize the timbre of an instrument. This source-filter structure provides a more semantic description of sound than additive synthesis: in certain cases the two blocks can be given a physical interpretation in terms of an exciting action and a resonating object, respectively. The source-filter technique has been used especially to produce synthetic speech, but it is also applied to musical instrument synthesis (Roads 1996).

The most common procedure in spectral modeling synthesis techniques is *linear predictive coding* (LPC) and speech synthesis. In this case, the broadband input signal can be interpreted as a glottal source signal, and the shaping filter represents the action of the vocal tract. Generally, LPC assumes either an impulse train or white noise as the input that is passed through a recursive filter (Markel and Gray 1976). Since the LPC model is parametric, the data obtained by the analysis has an exact interpretation in terms of the model. The linear transformation property has risen also in application to string instrument synthesis by periodic repetitions of input signal (Karplus and Strong 1983, Jaffe and Smith 1983). Avanzini (2001), for example, gives an illuminating up-to-date overview of speech synthesis.

Sinusoids plus residual synthesis techniques have proven to be useful for modeling and coding short musical tones. However, the problem in sinusoidal plus residual synthesis is that all forced oscillators, such as bowed strings, reeds, and voice generate nearly-periodically modulated noise, not additive noise. A better understanding of the physics of these oscillatory mechanisms (Rodet 1995) and new methods in higher order statistics (Brillinger and Irizarry 1998), as well as wavelets and time series (Irizarry 1998) are leading to better tools for multi-level decomposition of sounds into transient events, pitched and unpitched oscillations, convolutional noise and colored noise. These new models require efficient, real-time noise synthesis algorithms. Freed (1999), for instance, has proposed spectral line broadening, which is an example of such algorithm. *High Level Attributes* (HLA) model, where the most meaningful parameters for synthesis are extracted, has been discussed in (Jensen 1999). Some efficiency improvements are also done grouping similar partials obtaining *Group Additive Synthesis* (e.g. Kleczowski 1989).

The *phase vocoder* is a special technique that belongs to a family of sound analysis/synthesis systems. In spite of sound transformation properties, furthermore it allows sound synthesis from analysis data, and development of musical structures based on acoustical properties (Serra 1997). Typical analysis-synthesis system includes the phase vocoder, additive analysis-synthesis, formant analysis-synthesis, and linear prediction. The analysis-synthesis techniques, such as the phase vocoder are discussed also in section 2.2.

As been discussed earlier, *granular synthesis* has properties of many synthesis techniques. In some cases granular synthesis becomes the implementation of an inverse transform derived from time-frequency representations, such as the STFT or *wavelet transform* (Kronland-Martinet and Grossman 1991, Godsill and Rayner 1997, Vetterli 1992), and in periodic signal cases, *pitch synchronous granular synthesis* is obtained. Granular control to a form of subtractive synthesis has been applied by De Poli and Piccialli (1991).

The *formant wave-function* (*Fonction d'onde formantique*, FOF) synthesis (Rodet 1984) is based on the idea of using an excitation signal and a filter. The formants defined by their center frequencies, bandwidths, amplitudes, and envelopes are extensively used for synthesis of speech and singing (Roads 1996). The frequencies of the tones used in the system are readily determined by the periods of the excitation train pulses. In general, the response of the filter can be interpreted as a sum of responses of a set of parallel filters, each of which corresponds to a formant in the synthesized sound. An FOF synthesizer is constructed by connecting FOF generators in parallel, which can be controlled via the CHANT program (Rodet *et al.* 1984). The program was originally designed to produce high-quality singing synthesis, but it can also be applied for musical instrument synthesis.

Voice simulation (VOSIM) is a method of creating synthetic sound by trains of pulses of a simple waveform developed by Kaegi and Templaars in 1978 (Roads 1996). The pulses have a variable duration and delay, and they are shaped like squared sinusoids. The method uses external parameters for vibrato and transitional sounds, for instance.

Physical Modeling

The physical interpretation opens a way to capture valued aspects of real instruments, which have been difficult to obtain by more abstract synthesis techniques. When the basic model is physically meaningful it is often obvious how to introduce nonlinearities correctly, thus leading to realistic behavior beyond the reach of pure analytical methods.

In the context of physical modeling one can emphasize also the terminology to be used about the sources of sound. In sound source modeling the goal is to obtain a representation of a sound in terms of models of the sources that produced the sound. Depending on the application, sound source modeling techniques may be closely related to the synthesis side of the framework.

The sound source modeling typically starts with a study of the fundamental physical behavior of the sound source, for instance, a musical instrument. The laws of physics that govern the sound production mechanism of the instrument are expressed in terms of mathematical equations and relations. In the second step, a computational model is devised based on the expressions derived above. Depending on the application, computational efficiency or accuracy of the simulation is emphasized in the model. When a computational model has been implemented, it is calibrated by determining its parameters so that the simulation is as authentic as possible. This typically involves analysis and comparison of tones produced by both the virtual and the real instrument.

Based on the discussion above, the dynamically coupled systems of the musical instruments can be governed either by the linear or by the nonlinear wave equation. The traveling wave solution of the wave equation was first published by d'Alembert in 1747 (Smith 1997). First notes of traveling wave solutions applied in real time were given by McIntyre, Schumacher and Woodhouse (1983). Smith (1983) presented a computationally highly efficient model.

The dynamical properties, together with the computational constraints mentioned above, favor the time-domain modeling techniques over their frequency-domain counterparts. There are two main approaches in time-domain model-based sound synthesis: *numerical integration* and *digital waveguide synthesis* (Smith 1992) that is based on the traveling wave solution of the linear wave equation.

Digital waveguide models (DWG) are essentially discrete-time models of *distributed* media such as vibrating strings, bores, horns, and plates. They are often combined with models of *lumped* elements such as masses and springs. Distributed model implementations typically consist of delay lines, or DWGs in physical modeling, in combination with digital filters and nonlinear elements. In digital waveguide models, distributed losses and dispersion are still summarized and lumped at discrete points as digital filters, separating out the pure delay-line which represents ideal propagation delay. Distributed waveguide models can be freely combined with lumped filter models; for example, a brass instrument model typically consists of a lumped model for the "lip reed" and a distributed waveguide model for the horn (Smith 1996).

The *digital waveguide synthesis* (DWG) method relies on the system, in which the interactions between modeled elements operate at individual locations, so that the rest of the system can be modeled efficiently by bidirectional delay lines, i.e., digital waveguides (Smith 1992). Digital waveguide models are more closely related to unit element filters, which were developed much earlier in microwave engineering (Smith 2002). The DWGs and WDFs (see above) have sometimes been confused. However, a digital waveguide model may be obtained via simple sampling of the traveling waves in a unit-element filter, whereas a wave digital filter is typically derived via the bilinear transformation (Smith and Abel 1999) of the scattering-theoretic formulation of an RLC (or mass-spring-dashpot) circuit, with some additional special techniques for avoiding delay-free loops (Smith 2002).

DWGs have been extensively applied in plucked string synthesis. Dispersion caused by string stiffness has been simulated using all-pass filtering (Smith 1983, Van Duyne and Smith 1994). Parameter calibration methods based on STFT have been studied (see section 2.2). In other musical instruments, a slapbass synthesis (Rank and Kubin 1997) and piano synthesis (Smith and Van Duyne 1995), as well as clarinet (Smith 1986, Välimäki *et al.* 1992b) and flute (Välimäki *et al.* 1992a, 1993, Välimäki 1995) syntheses have been presented.

Digital waveguide filters (DWF) have proven useful for building computational models of acoustic systems, which are both physically meaningful and efficient for digital synthesis (Smith 1987). The waveguide filters have been used as tools to build reverberators (Smith 1985), to model computationally efficiently bore of a wind instrument and nonlinear coupling between violin bow and string (Smith 1986). The DWF building blocks feature exact physical interpretation of the contained digital signal as pressure or velocity waves, for instance. Furthermore, the signal power is available instantaneously with respect to both space and time. On the other hand, one of the fundamental benefits is that the DWFs can also be reduced for realizing any digital filter transfer function.

DWFs are also very closely related to *Wave Digital Filters* (WDF), which have been developed by Fettweis (1986). The main focus of the method is on using lumped models. For realizing the lumped models with feedback, wave digital filters also incorporate short, unit-sample, waveguide sections called unit elements, though being additional to the main development. WDFs can also be viewed as a particular kind of *finite difference scheme* having unusually good numerical properties (Bilbao 2001). Multidimensional properties of both WDFs and DWGs have been extensively studied in Bilbao (2001). In an elaborated model of the piano (Van Duyne *et al.* 1994), a wave digital hammer employs a traveling-wave formulation of a lumped model being therefore analogous to a WDF.

The cost in efficiency of the time-domain finite difference model (see below for more details) in more than one dimension is one of the motivations behind the research on *digital waveguide meshes*, where the DWG theory has been extended to higher dimensions (Van Duyne and Smith 1995, Savioja 2000).

Historically, physical models of musical instruments led to prohibitively expensive synthesis algorithms (e.g. Chaigne and Askenfelt 1994) analogous to articulatory speech models. These time-domain techniques are sometimes referred to as "brute force" methods, since they are based on direct discretization of the *partial difference equations* (PDE) and have high computational costs. Numerical integration methods approximate the wave equation by replacing the partial derivatives either by finite differences, or mechanical elements like masses, springs, and dampers (Cadoz et al 1983). Historically, the *time-domain finite difference* (FDTD) approximation provided the first physical modeling synthesis scheme (Hiller and Ruiz 1971). Finite difference scheme has been used for modeling idiophones generally for analysis and non-real time purposes (Chaigne and Doutaut 1997, 1998) and, for instance, single-reed instruments (Stewart and Strong 1980, Sommerfeldt and Strong 1988) and string instruments such as the guitar (Chaigne 2002, Karjalainen 2002). The FDTD techniques have been extended to analyze struck strings of piano (Chaigne 1994), vibrating bars and resonators of a xylophone (Chaigne 1997, 1998), and damped impacted plates (Chaigne and Lambourg 2001, Lambourg *et al.* 2001). A considerable computational power is usually needed in the FDTD simulations. Usually, two or three-dimensional FDTD simulations cannot be run in real-time.

The *modal synthesis* has been developed at IRCAM, France (Adrien 1991). With a commercial software Mosaic (Morrison and Adrien 1993), developed further to Modalys (Eckel *et al.* 1995), the user can simulate vibrating structures using the principles of modal synthesis. The basis of the method is that any sound-producing object can be represented as a set of coupled vibrating sub-structures, which are defined by modal data (Adrien 1991). The coupling provides the energy flow between the structures. Typical sub-structures to be modeled are membranes, plates, tubes, bows, bells, and bodies.

First real-time model-based sound synthesis systems were the *CORDIS* (Florens and Cadoz 1991) and the *CORDIS-ANIMA* (Florens and Cadoz 1991, Roads 1996), the latter being developed for simulation of two and three-dimensional elements. The system describes vibrating bodies as a set of interconnected mass-spring-damper cells. One disadvantage of this and the previous, modal synthesis approach, is high computational cost. Furthermore, no analytical tools are available for assessing the stability properties of the discretized systems.

Recently, some new interest and efforts towards physically-based simulations joined to interactive motion, and graphics has been gained. For instance, van den Doel *et al.* (2001) introduced the FoleyAutomatic, where algorithms for real-time physically-based sound synthesis for interactive and multimedia have been presented.

Extended Karplus-Strong models

A physical modeling interpretation of the Karplus-Strong algorithm has been taken for the first time by Smith (1983) and Jaffe and Smith (1983). *Single delay loop* (SDL) models (Karjalainen *et al.* 1998) are an extension to KS models of today that need to be distinguished from both the non-physical KS algorithm and the bidirectional DWG models. The SDL model is used extensively in application to plucked strings modeling, as discussed in the following section (commuted waveguide synthesis).

Commuted Waveguide Synthesis

Commuted Waveguide Synthesis (CWS) was first introduced by Smith (1993) and Karjalainen *et al.* (Karjalainen and Välimäki 1993, Karjalainen *et al.* 1993). In the CWS model, the principle of commutativity of linear and time-invariant elements is employed. The commutative elements are such as body and string of an instrument. The response of the instrument body can be incorporated in the input signal together with the excitation[7]. One of the first non-linear CWS models was the kantele synthesis (Karjalainen *et al.* 1993). The commuted synthesis is discussed extensively in application to the guitar by Karjalainen *et al.* (1993) and Välimäki *et al.* (1996), with multi-rate extensions in body resonators (Välimäki and Tolonen 1997) and structures for coupling model of guitar strings (Tolonen *et al.* 1998). More recently, linear DWG plucked string models have been extended to simulate non-linear traveling waves (Tolonen *et al.* 2000) and ethnic plucked strings, such as the ud and the lute (Erkut *et al.* 2001), as well as the Turkish tanbur (Erkut 2002). The models discussed in these papers utilize an extended Karplus-Strong algorithm. A commuted waveguide piano model including a linearized piano hammer was described by Smith and Van Duyne (1995) and Van Duyne and Smith (1995). In application to the violin, the commuted synthesis was discussed by Jaffe and Smith (1995). The commuted synthesis in application to the violin body modeling has been discussed in paper I and section 3.2.2.

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[7] The excitation varies depending on the instrument being, for instance, a pluck in the guitar, bowing in the violin, or a hammer strike in the piano.

3 PHYSICAL MODELING AND MODEL-BASED SYNTHESIS OF THE VIOLIN

The *digital waveguide modeling* (DWG, see section 2.3) is perhaps the most efficient method for physics-based modeling of musical instruments. In this thesis the model of the violin is based on the DWG methods. For this reason, discussion an DWG modeling is emphasized in this section.

Earliest mathematical and physical theories of bow-string interaction have been described by McIntyre and Woodhouse (1979), McIntyre, Schumacher and Woodhouse (1983), and Cremer (1984). Resting on these theories, the first physics-based approach to use digital filters to model a musical instrument has been taken by Smith (1983) in application to the violin.

For physical modeling and model-based sound synthesis, it is necessary to know something about the different parts of the instrument to be modeled, such as linear and non-linear behavior both in the parts and in the instrument as a whole. Also, a crucial task is to know how the total system of the musical instrument works, and how the different parts couple to each other finally producing the sound. After understanding how the system works in general, it is possible to utilize digital techniques to simulate the phenomena.

The following sub-sections present an overview of the violin and its simulation. Firstly, the functioning of different parts of an instrument is discussed and general theories are introduced. Secondly, the violin simulation is carried out by means of physical modeling sound synthesis in sub-section 3.2.

3.1 Overview on the violin

The violin is a bowed string instrument, which shares the control characteristics of most of the other instruments in this family. Bowed instruments include the violin, the viola, the cello, and the contrabass. The violin gives an appearance of deceptive simplicity to the eye, but is in fact constructed of some 70 parts, which require the skill of a master craftsman to cut and assemble.

Generally, the violin is a plucked or a bowed instrument. In the first case, the sound and vibration amplitude follows a predefined envelope, whereas in the latter case, the amplitude is continually controlled by the bow. Likewise, the pitch and different timbre dimensions can be defined as being a function of the movement of the bow, and the fingers of the left hand.

The violin is tuned G3, D4, A4, and E5. The range of each string is about 2 octaves. The violin has the following parts, which are of interest to us: strings including fingerboard, bridge, and body. The bow is constituted of bow hair, and the hair length is about 65 cm. Sound is produced when the bow touches the string. The string often continues to vibrate after the bow no longer touches it, thus producing an envelope-based sound.

Any violin has a certain potential of sound volume, whose realization depends partly on different parts of the instrument, such as the type of strings and their tension, the type of bridge, the quality of the bow, and partly on the skill of the player. Fingering, vibrato, bow speed and pressure, and the relative placing of the bow between the bridge and the end of the fingerboard all have a direct bearing on the dynamic and tonal characteristics of the sound (Fletcher and Rossing 1998, Boyden and Walls 2002). These properties are the key parameters to be simulated in the physical modeling of the violin.

When the bow sets the string or strings in motion, the vibrations are transmitted to the soundboard and the back via the bridge and the soundpost that renders the right foot of the bridge effectively immobile, leaving the left one relatively free to transmit vibrations to the bass-bar and soundboard, and thence, through the sides to the back, whose primary function, however, is that of a reflector. The total area of the instrument body resonator then further amplifies the vibrations and transmits them eventually to the ear of the listener. The soundholes in the body operate as a secondary and complementary acoustical system, adding considerably to the resonance. (Hutchins 1997a, 1997b, Boyden and Walls 2002)

The quality and character of the tone depend on the vibrating string and how well its fundamental pitch frequency and upper partials are received and transmitted by the wood of the violin's body. The string vibrates not only as a whole but also in various parts of its length so as to produce the other harmonics of the fundamental, thus giving richness and complexity to the timbre (e.g. Fletcher and Rossing 1998).

History of the violin

The classic outline of the violin became standard in Italy around 1550, which was a period of rapid change and adventurous experimentation in instrument making. It has been investigated, that in the 15th century there have been two main types of bowed instrument: the alto-range medieval fiddle, usually with five or more strings, and the small pear-shaped rebec, with two or three strings (Boyden and Walls 2002). The instruments have had a flat bridge or no bridge at all, which means that they must have been used essentially to play monophonic music in chords. By the same manner the Greek *lira* and the Norwegian

Hardanger fiddle are still played today (Boyden and Walls 2002). A bit later, a bowed *viola* emerged with strings stretched in a protuberant manner so that the bow can touch any one string the player wills, leaving the others untouched.

The viol and violin families seem to have been developed as part of a humanist cultural agenda that preferred 'noble' strings to 'ignoble' winds. String instruments are consistently associated with virtue, spiritual love, and harmony, while wind instruments are associated with vice, sensual love, and strife. The viol, soft, sonorous but rather lacking in attack, was suitable for serious contrapuntal music and for accompanying the voice, while the sprightly violin was quickly recognized as the ideal vehicle for the new composed polyphonic dance music that developed soon after 1500. With Andrea Amati and his sons the center of Italian violin making moved to Cremona. (Boyden and Walls 2002)

Most violins made before 1800 were modified to yield greater tonal power and brilliance. The Stradivari with flat-shaped bodies flourished as concert instruments, while the highly arched, smaller-toned Stainers and Amatis lost their former popularity. The main body of the violin remained unaltered despite further attempts at 'acoustical improvement', ranging from the construction of instruments from metals, glass, leather, plastics, and ceramics to experiments with various shapes, notably Félix Savart's trapezoidal violin with straight sides and straight slits for soundholes in 1817, and François Chanot's guitar-shaped model with small crescent soundholes. (Boyden and Walls 2002)

During the 1960s and 70s, Carleen Hutchins and her associates developed a 'concert violin', with a longer, revamped body and larger f-holes, and a string octet, compromising mathematical, acoustical, and violin-making principles to produce instruments in different frequency ranges that possess the dynamic power and timbre of the violin family. These instruments range from the contrabass violin to the small treble, or sopranino, a violin tuned an octave above the normal violin (Hutchins 1997, Boyden and Walls 2002).

Experiments with building electric instruments based on the violin family, often with solid bodies and amplified usually by means of one or more sets of electromagnetic pickups or contact microphones, have continued since the 1920s. By allowing the instrument's signal to be amplified, modified or altered through changes in frequency response, rapid changes in amplitude, harmonic alteration, echo and reverberation effects, and distortion, the violin has served a wide range of classical and popular musical styles. (Boyden and Walls 2002)

Sound production and acoustics

The role of the violin body is to amplify and project the string vibrations to the outer air (Fletcher and Rossing 1998). What makes a particular violin good is the degree to which it transmits the string vibrations of the fundamental and its harmonics with equal response over the whole register of the instrument (e.g. Hutchins 1997b). The tone of the violin, then, depends initially on the capacity of the many resonance frequencies of the wood to respond to the string vibrations (e.g. Hutchins 1981, Jansson *et al.* 1988). Many makers, when adjusting the final thicknesses of the back and belly of a new violin, tap the plates to tune them. The notes produced are known as ‘tap tones’. The natural resonance of the interior air space – the so-called ‘air tone’ – has a frequency normally in the area of the D string in superior violins (e.g. Hutchins 1981).

Many experiments have been made, especially in the last 50 years, to determine which factors affect the timbre of a single note or of all the notes of a particular violin, thus distinguishing one violin from another (e.g. Benade 1987). Modern acoustics, using electronic equipment, has shown that new theories in musical instrument acoustics need to be introduced (Hutchins 1997a). There are still major questions regarding the acoustics of the violin that are not yet completely or satisfactorily answered – for example, what makes a violin a ‘good’ one, and whether old violins are better than modern ones (Benade 1987).

Since its origins, the violin has undergone a considerable evolution of detail to meet the changing requirements of successive generations of performers and composers. The first century and a half of the ‘true’ violin culminated in the magnificent ‘classical’ model of Antonio Stradivari shortly after 1700. This was not the end of the instrument's evolution; in the early years of the 19th century it was altered in a number of respects to attain greater power and a more mellow tonal quality. In this era the Tourte bow also gained universal acceptance. (Boyden and Walls 2002)

Whether we regard these changes as improvements is an entirely subjective matter. Many musicians now take the view that a particular repertory will be served best by performing it with instruments set up and played in the way the composers of the time expected. It is in response to this approach that so many violinists have now acquired ‘Baroque’, ‘Classical’, or even ‘Renaissance’ violins.

3.1.1 Bowed string

Bowed strings, being highly nonlinear musical instruments by their mechanical behavior, have been an intriguing challenge for researchers for over a century. To be able to create models of bowed strings it is crucial to have an understanding about the physical behavior of the real bowed string. The goals of this activity include obtaining the underlying physical knowledge of the real instrument and developing a sufficient model of it, with respect to both mathematical methods and musically satisfying sound synthesis.

Modeling of bowed strings has a long tradition. It is a well known fact that the audibly most significant part is the vibrating bowed string itself, rather than, for instance, the sound radiating instrument body, although the latter has an important effect on sound coloration. The phenomenon of the bowed-string vibration was investigated for first time over a century ago (Hutchins 1997). Later, the examination of bowed strings has been divided to several branches, such as physical modeling, simulation and model-based sound synthesis. To date precise models have been developed and promising sound quality in synthesis obtained (see section 3.2), although several aspects in model-based sound synthesis still need to be examined.

The violin string is a complicated linear system, which is set into self-excited vibration by the action of stick-slip friction between the bow and the string. In the early days the theory of bowed string excitation assumed that the string and the bow hair move and stick together during the forward motion of the string, while during the return slip the friction is negligible. History of investigations on string vibration in the violin family may be seen as started by Duhamel, who recognized the idea of stick-slip –periodicity of bow-string interaction (Hutchins 1997). After this Helmholtz and his contemporaries continued the work on bow-string interaction with findings of, for instance, stretching of the string and the Helmholtz corner in the string motion (see e.g. Cremer 1983, Fletcher and Rossing 1998, Fig. 1 in paper VI). The center point of bowing as a fraction of the total string length L is commonly denoted by β. Normally, when playing a violin, β has a value of 0.3 at maximum.

Later, work on bowed strings has been carried out by McIntyre, Schumacher, and Woodhouse. The dynamics of bowed strings and their aperiodicities are discussed by McIntyre and Woodhouse (1979), Schumacher (1979), and McIntyre *et al.* (1981). The modeling of the motion of bowed string has been investigated and described in good detail by McIntyre and Woodhouse (1979), McIntyre *et al.* (1981, 1983), and Woodhouse (1993a, 1993b). In the model discussed by Woodhouse (1992) the two relevant variables to consider at the bowing point are the frictional force between the bow and the string, and the transverse velocity of the point on the string beneath the bow. The models that are most commonly studied often make idealizations about the physics of the real system. In bowed string models, torsional waves need to be considered, since their presence affects bow-string dynamics. The effect has been studied by McIntyre, Schumacher and Woodhouse (1983). The attack and the transient on the bowed string as well as anomalous low frequencies have been studied by Guettler (2002).

When modeling bowed strings, the bow-string interaction is typically approximated at a single point. In real playing, however, the finite width of the bow is known to have an effect on the sound produced. One can, for instance, change the sound quality by tilting the bow, which changes the number of bow hairs that are in contact with the string. The major result of finite width bowing, aperiodicity in bowed-string motion with presence of noise, has been shown by McIntyre, Schumacher and Woodhouse (Schumacher 1979, McIntyre and Woodhouse 1979, McIntyre *et al.*1983) to be caused by sharp spikes superimposed on the Helmholtz sawtooth. The major cause of these spikes is a mechanism depending on the finite width of the bow hair in contact with the string (Hutchins 1997a). McIntyre *et al.* (1981) observed a so-called differential slipping of the contact of the bow and the string during the sticking phase. The basis of the theory of bowed string was established by Benade in 1987. Preliminary studies on modeling of the finite width of the bow in continuous time domain have also been done (McIntyre *et al.* 1981, Pitteroff 1993, Pitteroff and Woodhouse 1998). Finite width bow-string excitation has been investigated in application to physical modeling sound synthesis by the author in III and VI.

Pickering (1985, 1986) has measured and reported rotational modes of real strings. The nonlinear energy transfer of different frequency modes on a vibrating string has been investigated both theoretically and experimentally by Legge and Fletcher (1984). A digital bow on a real string has been used for more accurate measurements of the frictional characteristics of the bow-string interaction (Weinreich and Caussé 1986) and a bow controller has been used for examination of the bow strokes (Serafin and Young 2003).

3.1.2 Bow

The style of the violin bow has changed over time from baroque era until today. First bows were arched upwards, whereas the modern bow is arched downwards. Drastic change in sound loudness and in player's pressure control was obtained by this improvement (Hutchins 1997b, Boyden and Walls 2002).

The traditional source of bow hair are the tails of white horses, but some players use black horsehair or synthetic substitutes such as nylon, arguably for a coarser tonal effect. Huberman has made experiments with fine-gauge wire which also yielded mixed tonal results (Boyden and Walls 2002).

Askenfelt (1986) has developed a method to record the motion of the bow transverse to the strings and the downward force on the string exerted by the player under actual playing conditions (Hutchins 1997a). Furthermore, the method has been used to include measured parameters of bow position and velocity, bow-bridge distance, and bow force, which have further been compared with player's use and prediction of changes in parameter values (Askenfelt 1989).

A preliminary study about a method to measure properties of bow dynamics, including both longitudinal and transverse bow hair motion, has been presented by Schumacher (1975). The model has been incorporated into a theoretical behavior of bow hair motion by Schumacher (1979). The string rotational motion has been incorporated with bow dynamics by McIntyre *et al.* (1983).

Experimental techniques to measure bow hair dynamics have been used by Bissinger (1993). He combined modal analysis measurements of accelerometer modes on a violin bow stick with microphone modes of the bow hair obtaining composite modes for overall bow hair motion.

3.1.3 Body

Probably the most important determinant of the sound quality and playability of a string instrument is the vibrational behavior of its body. The vibration has been studied already for 150 years, though the greatest understanding has been reached during the last six decades. Nowadays, the optical holography, electronic measuring devices, and computers have greatly contributed to the understanding of the behavior of the vibrating body.

The eigenmodes of a violin, as the normal modes of vibration are determined mainly by the coupled motions of the top and back plate, and enclosed air. The other parts of the instrument, such as the ribs, neck, and fingerboard seem to have smaller contribution in the modes. Different researchers seem to have used different systems of nomenclature for the various vibration modes in violins. The distinction between the systems of nomenclature for air, top and body modes, by Jansson and his co-workers, and the system used by Hutchins and her co-workers is discussed by Fletcher and Rossing (Fletcher and Rossing 1998).

With different analysis methods good approximations and estimations of preferred mode frequencies have been obtained and some of them are summed by Fletcher and Rossing (1998).

Body analysis

Investigations on the acoustical properties of the violin body vibrations have been carried out with quite a number of different experimental methods. Mode frequencies appear to differ widely between different instruments, and the agreement between the results of different investigators is not always good.

Félix Savart was the first researcher who investigated the modes of the complete violin (Cremer 1983, Hutchins 1997a). The first researcher working with electronic test equipment, such as the oscilloscope, the heterodyne analyzer, and recording devices, was Backhaus in the 1920's and 30's (Fletcher and Rossing 1998, Hutchins 1997a). He used capacitance sampling and search-tone analysis, and vibrated the violin with an automatic bowing device at the main resonance frequencies (Hutchins 1997a). The work of Backhaus was carried on by his student Meinel, who investigated the possibility to improve the behavior in a particular frequency range by removing wood from a specific area (Meinel 1957). The only study so far on the vibrational modes of the total instrument was done in 1950's about the cello by Eggers (Cremer 1983). He used a capacitive pick-up and conductive paint, and an electro-dynamic transducer driving a rod attached to the string near the bridge. Later, Menzel and Hutchins (Fletcher and Rossing 1998) used an optical proximity detector. Luke (Luke 1971) scanned the plates with optical sensors applying a sinusoidal force to the bridge.

With a new technique, the optical hologram interferometry in the late 1960's, more detailed and comprehensive information in the study of violin body resonances was obtained by Reinicke and Cremer (1970) and Jansson *et al.* (Fletcher and Rossing 1998), where the location of nodal lines were recorded accurately. Visualizations of transient response with different excitation methods have been discussed by Molin *et al.* (1990, 1991). More refined technology was used in the 1990's when real-time electronic holography was deviced (Jansson *et al.* 1994).

The first modal analysis[8] for the complete instrument has been carried out by Müller and Geissler (1983), Jansson *et al.* (1986) and by Marshall (1985), where the large-amplitude motions in the neck and fingerboard are suggested to correspond to the feeling in the hand of a player in lower-order modes. The modal-analysis animated video recording gave visualization of amplified bending motions. Impact excitation allows investigation of the motion at a number of frequencies at one time, and it is possible to compute the modal parameters of interest, including modal frequencies, damping, and mode shapes. The modal analysis has been used also with other musical instruments (Fletcher and Rossing 1998).

Modal analysis of the mode shapes and corresponding frequencies has been done for free plates considering the wood stiffness (Rodgers 1988), the local thickness changes on free-plates (Rodgers 1990a) and on top-plates (Rodgers 1991b), as well as for arching of the plate and back-plate (Rodgers 1990b).

Finite element analysis as a companion to modal analysis provides another tool to investigate the vibration modes of the violin. The technique has been applied to unattached violin plates (Roberts 1986a, Rodgers 1988, 1991a). The finite element method has been used in modal analysis of violin. The first achievement in stringed instrument acoustics was in 1974 with Schwab's finite element description of the modal shapes of a guitar top plate (Knott 1987). The first full finite-element analysis of a violin from free plates to finished instrument was done by Roberts (Roberts 1986a) using *NASTRAN* (see Section 2.2). The work on guitar plates was continued by Rodgers, Rubin and Farrar, and Richardson and Roberts (Knott 1987, Fletcher and Rossing 1998), and the finite element method was applied to the study of violin plate vibrations.

The *MSC/NASTRAN* finite analysis program was used together with graphics processing language *PATRAN* (Knott 1987). In the analysis, a violin structure consisting of over 2.000 elements with 10.000 degrees of freedom with certain material parameters (Haines 1979) is modeled and the displacement graphics of the mode shapes of both the free plates and the completed violin are provided, and furthermore compared with the results by Hutchins (1981) and Marshall (1985). In modal analysis also the neck, tailpiece, and bridge have been added into the body model (Knott 1987). Research with varying damping and coupling properties was carried out by Rodgers (Rodgers 1991b).

Studies of free top and back plates in the 1950's were conducted by Saunders and Hutchins (Hutchins et al 1960). Schelleng mapped the vibrational modes up to 600 Hz with magnetic devices (Hutchins 1997a). At about the same time, Stetson suggested the use of hologram interferometry and the method for testing both free plates and complete violin (Hutchins 1997a). To make the modal patterns visually clearer, Hutchins developed a modified Chladni[9]

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[8] Modal analysis is the process of describing the dynamic properties of an elastic structure in terms of its normal modes of vibration. Studies on modal analysis generally deal with either mathematical modal analysis or modal testing.

[9] In 1787 Chladni demonstrated a method where he dusted a surface with fine sand or the like, and excited the surface at various frequencies. When the surface was excited

pattern technique (Hutchins 1997a) that was experimented with various cases of plates with other parts, such as soundpost and the bridge, attached to the finished instrument (Hutchins 1981, Moral and Jansson 1982a). Some quality aspects of the modes have been discussed in comparison tests, in which some of the modes have been preferred to the other (Moral and Jansson 1982b, Hutchins 1991).

Thompson has studied the effect of humidity, different covering, and the age of wood (Thompson 1979). The correspondence of elasticity parameters of wood in the mode frequencies has been investigated by Molin, Lindgren, and Jansson (Molin et al 1988) and the effect of the thickness of the plates by Jansson, Moral, and Niewczyk (Jansson et al 1988). The latter have also studied the plates with different boundary conditions (Jansson 1989). Several investigations have compared violins with and without a soundpost, and the results pointed out that removing of the soundpost caused lowering on the mode frequencies and decrease in the radiation efficiency (Saldner 1996, Bissinger 1995).

The properties of sound radiation of the violin body are investigated also by using the *Linear Maximum Length Sequence* (MLS) technique (Farina et al 1995) in application to measurement of ancient violins (Farina et al 1998). In this technique, the body response is treated as a numerical filter to convolve with excitation signal.

3.1.4 Other parts of the violin

The behaviors of torsional vibrations (Schelleng 1973, Cremer 1984) and rosin friction (Gillan and Elliot 1989, Smith and Woodhouse 2000) have been preliminarily studied. Some of the results have been converted into digital signal processing models (Smith 1983, Serafin *et al.* 1999, Serafin and Smith 2000). In some studies the impact of torsion is included in the friction model by shearing the friction curve (McIntyre *et al.* 1983), but because the torsional vibration is a more refined process, it needs to be handled separately, as done in the paper of the author (VI).

The torsional properties of a real bowed string have been investigated or discussed by only a few researchers (Schelleng 1973, McIntyre *et al.* 1981, 1983, Cremer 1984, Caussé and Weinreich 1989, Woodhouse *et al.* 2000). Some of the experimental setups have been criticized by Woodhouse (1992). However, the obvious importance of torsional motion in the bowed string model has been acknowledged, as discussed by Woodhouse *et al.* (2000). Already Schelleng (1973) provided useful information about the ratio of torsional to transverse impedances and the torsional wave speeds.

The dynamical properties of violin bridges have been extensively studied by Cremer (1983). The discussion of the soundpost and the bassbar has often been incorporated in the investigation of the top and back plates of the

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at a natural frequency, the grains migrated to parts of the surface that were at rest, to the nodal lines (Cremer 1984).

instrument. Hutchins (1997a, 1997b) has discussed the effect of these and the tailpiece of the instrument.

Weinreich (1993) has carried out an intriguing study about what science knows and doesn't know about violins. In the study some fundamental questions about the state of physics in the research of bowed strings are discussed, for instance, from the point of view of internal dynamics of the bow and the string as well as of a sound radiator. In the study the secrets of Stradivarius have also been discussed, revealing the criteria that judge the most excellent sounding violin in the world.

3.2 Simulating the violin

In the past few decades the theory of bowed strings has been developed to permit realistic computer simulation of the dynamics of real strings bowed by real bows.

The goal in sound synthesis of today is to provide the most cost-effective computation for a given quality level. Of great interest is to determine the relative value of the various model components for the synthesis, for both the sound and the playability (see section 4) part of the virtual instrument (see section 5). Questions of this nature can be discussed using computer-based time-domain simulation models.

A practical implementation of sound synthesis can be done by using digital signal processing techniques and filtering tools (Ifeachor 1993, Orfanidis 1995). A thorough introduction to the background and innovations on digital signal processing has been given by Rabiner and Rader (1972). When modeling and simulating bowed strings, it has been common to intentionally leave out some aspects of the physical process, in order to obtain a better understanding of the actual physical phenomena. To simulate the process with higher accuracy it would be, however, crucial to include these complex physical properties, at least on a moderate level by using physical model based DSP methods.

From the technical point of view the amplitude of the sound is dependent on the speed of the bow, the force (pressure) of the bow, and the position of the bow on the string. The amplitude of the violin is roughly proportional to the force of the bow. This force has a maximum and a minimum value, which are functions of the position of the bow (Schelleng 1973, Fletcher and Rossing 1998). The onset of the sound also changes with the force and acceleration (Guettler 2002) of the bow. The simulation of the performance of bowed string models has been discussed in section 3.3.

Commuted waveguide synthesis, as discussed earlier in physical modeling section 2.3, is a computationally highly efficient method to model also bowed strings (Jaffe and Smith 1995). In this approach the bowed string is modeled as a periodically plucked string. Commuted models that are based on resonator impulse response implemented in the excitation signal have advantage over many other physical models, instead of expensive implementation of explicit models of different parts of the instrument.

Commuting the string and the resonator means that the string is plucked by a *periodically repeated resonator impulse response.* A nice simplified vibrato implementation is obtained by varying the retriggering period of impulse-response so that the vibrato is implemented in the oscillator of excitation. The string loop delay needs not be modulated at all. While this synthesis method departs from being a physical model, the vibrato quality is satisfying and qualitatively similar to that obtained by a rigorous physical model.

In addition to plucked and struck strings discussed in the previous section, simplified bowed strings have been calibrated to recorded data using LPC (Smith 1983, 1993). In this model, the bowed string is approximated as a periodically plucked string. Commuted synthesis of the violin is discussed by the author in application to body modeling in I.

3.2.1 Bowed string synthesis model

The aim of the theory formulation of bowed strings has been to allow computer simulations of the dynamics of real strings bowed by real bows. This indicates that it is possible to calculate the properties of real strings and their effects on the dynamical properties (Hutchins 1997a). One of the first detailed descriptions of bowed strings combining the non-linear behavior between the bow and the string capable of predicting plausible behavior was introduced by McIntyre and Woodhouse (1979). At the same time, Schumacher (1979) calculated the bowed string motion mathematically, providing some physical effects in practice. Later, Guettler (2002) has done a computer simulation of some of the characteristics of the bowed string attack transients relating to torsion and transverse motion, frictional losses, and the effects of bow compliance. Woodhouse has explored the playability of violins. In (Woodhouse 1993a), a theoretical framework on the differences of playability by the reflection functions has been considered, and in (Woodhouse 1993b) the framework has been employed to study the problem of playability, also by computer simulation. The simulation of the violin bow (see section 3.1.2) is typically considered in addition to the string model, thus being integrated as a bow-string model.

The general solution of the wave equation for a string is composed by two independent transversal waves traveling in opposite directions (see e.g. Fletcher and Rossing 1998). In efficient physical modeling the string is described as an ideal lossless waveguide (Smith 1993), and the system of one string of a violin may be modeled with a delay line pair and filters in string terminations. One of the first implementations of string models has been given by Jaffe and Smith (1983). Smith (1986) used the DWFs in application to model the nonlinear interaction between the violin bow and string. Digital sound synthesis algorithms have been developed based on the models discussed in section 3.1.1 (Smith 1996, 1998).

Filters describing the losses at the string model terminations may often be combined into a single filter, also called a loop filter in some applications (e.g. Karjalainen *et al.* 1993, Välimäki *et al.* 1996), though a model of a finger pressing the string at some position on the fingerboard of the instrument needs to be considered as a separate filter. The violin model is described and illustrated in the papers of the author (II-VI).

An ideal string, which is a good approximation of a long, thin string under high tension, can be modeled as a digital delay line that allows any information to travel back and forth the string as undisturbed waves that are further reflected back at string terminations. In the real case, however, the wave reflected from the termination is not similar to the arriving one. For a realistic, non-rigid termination (e.g. player's finger), the incoming impulse is smeared out in the reflected wave (McIntyre and Woodhouse 1979). This is most conveniently dealt with by using low-order FIR filters of varying size, depending on the type of termination. The fine-tuning part of the string model is handled by fractional adjustment of delay units (e.g. Laakso *et al.* 1994). In physical modeling sound synthesis the most commonly used waveform in digital delay lines is the velocity waves, though for instance the propagation of pressure, and acceleration or displacement of waves can be simulated as well.

The desired pitch cannot be realized with integer-sized delay lines, hence the *fractional delay* (FD) approximation is needed for fine-tuning the length of the string model. It is often desirable that the length is adjustable during the synthesis, for instance for glissando and vibrato effects. A comprehensive guide to fractional delay filter design has been given by Laakso *et al.* (1994). A good FIR type of approximation is Lagrange interpolation, which has been applied widely in string models (e.g. Karjalainen *et al.* 1993, Välimäki *et al.* 1996), and also by the author in the studies of this thesis.

Another implementation of the FD filter is applied in the contact point of the bow and the string. The efficient FD techniques have been used as a three-port scattering junction and successfully applied to the implementation of finger holes in woodwind instrument models (Välimäki *et al.* 1993) and plucking position of plucked strings (Karjalainen *et al.* 1993). In this thesis, the technique is applied in finite width excitation of bowed string. This *fractional delay finite width* (FDFW) model, subsequently also referred to as *fractional varying excitation point* (FVEP) model imitates the multiple bow-hair excitation at a finite region of a string. The models are discussed in papers III and VI.

The main vibration planes in strings are the transverse and the longitudinal vibrations. For instance, in plucked strings it is important to model the transverse vibrations in two planes, so that both horizontal and vertical positions are covered (e.g. Välimäki and Tolonen 1998).

One important aspect in higher quality synthesis is the sympathetic vibration between the strings. The principles of this coupling process have been discussed, for instance, by Jaffe and Smith (1983). In the model by Karjalainen *et al.* (1993) the coupling is simplified by feeding a small fraction of a string output to the other strings. A more refined model of guitar string coupling has been discussed by Tolonen *et al.* (1998). A coupling model for the violin physical model, following the principles of the study by Karjalainen *et al.* (1993), has been discussed and applied in II.

Typically, all bowed string models assume that the frictional force is simply a function of instantaneous sliding velocity with an approximately hyperbolic dependency (e.g. Fletcher and Rossing 1998). As stated by Woodhouse (1992), there is no experimental basis for this assumption. Only limited information about plastic friction characteristics has been obtained (Woodhouse 1992, Smith and Woodhouse 2000). Therefore, a synthesis model of rosin friction characteristics would be necessary. Rosin-type organic matter seems to have viscous behavior, so that the solid matter starts to melt rapidly at a certain temperature. It has been suggested that rosin achieves its frictional properties by partially melting at every slip and resolidifying at every stick (Woodhouse 1992, Woodhouse *et al.* 2000). On the other hand, Guettler (2002) summarizes that the friction properties obtained in the studies of Smith and Woodhouse (2000) lead, after all, to a hyperbolic dependency. The digital model used by the author follows similar physical tracks, and with the FVEP model discussed in VI, a good correlation to realistic frictional dynamics in bowed string simulation has been obtained.

3.2.2 Body Model

A comprehensive study of instrument bodies of the violin family, both qualitatively and quantitatively, is presented by Hutchins (1997a, and 1997b), and Fletcher and Rossing (1998), and the behavior of the violin body is discussed in section 3.1.3 of this thesis.

One of the first studies in modeling the string instrument body electronically has been successfully carried out by Mathews and Kohut (1973). In a reverberant body model the exact position and magnitude of a resonance is not as important as their decay times and distances from each other (Mathews and Kohut 1973).

Moral and Jansson (1982) have indicated that the vibrational properties of the violin are best described by single resonances below 1 kHz, and that above 2 kHz the properties of the body are determined by the density of resonances and their damping. The idea of adjustable single resonances has been used in violin body modeling in paper I of the author.

From the viewpoint of real-time sound synthesis traditional simulation methods for string instrument bodies and soundboards have been and still are too slow. Still today we aim for simulation methods computationally as efficient as possible. Since available processor power is increasing the advantages of explicit body models such as parametric controllability start to draw attention to advanced body modeling.

The sound from the system of a string instrument can in most cases be accurately modeled as linear and time-invariant. A practical implementation for the body model can be carried out by using DSP techniques and filtering tools (Ifeachor 1993, Orfanidis 1995). A typical starting point for body modeling is to measure the impulse response of the instrument body, which can be described as a combination of exponentially decaying sinusoids. The analysis methods for body modes are discussed in section 2.2. Traditional filters, such as FIR and IIR, as well as warped filter structures of varying order have been studied as body models by Karjalainen and Smith (1996).

The radiation properties of violins have been the subject of several studies (see Section 3.1.3). In paper I of this thesis, the pressure field from radiating instrument body is measured in single location in one-meter distance and the measured data is used for synthesis purposes. For reproduction purposes also spherical speaker systems have been discussed (Wessel 1991, Roads 1996). Investigations in spherical space impulse responses combined with the creation of spherical multi-channel speaker arrays to model and manipulate the radiative timbral qualities of violins in traditional acoustic performance spaces have been done by Cook and Trueman (1998, Trueman and Cook 1999, Trueman 1999).

One of the most efficient approaches to computationally implement a body model in sound synthesis is to aggregate the body response into a string excitation wavetable (Smith 1993, Karjalainen *et al.* 1993). However, in the present study the emphasis has been on the ability to control the model parametrically. The second-order resonant filters modeling the most prominent low-frequency resonances have been used in application to a violin model by the author in I. A detailed overview of body modeling techniques has been given by Karjalainen and Smith (1996). Filter-based body models have been studied earlier for the guitar (Karjalainen et al 1991, Karjalainen 1996), for the piano (Laroche 1994), and for the violin (Smith 1983). Multirate extensions in modeling of body resonators have been discussed by Välimäki and Tolonen (1997) and adopted for the violin body model by the author in I.

In two-part body modeling the low- and high-frequency regions are modeled separately. Such method has been used in guitar body modeling (Penttinen 2001). The high-frequency range can be modeled in a more statistical fashion, while the low-frequency range is modeled more precisely. In two-part modeling the low- and high-frequency models can be summed after low- and high-pass filtering, respectively. For high-frequencies conventional reverberation algorithms (Väänänen *et al.* 1997, Gardner 1998) have been used. Also waveguide meshes have been used for the high-frequency region of body modes in application to the violin (Huang *et al.* 2000).

One computationally efficient approach is to implement body models in a frequency-warped domain. The order of a warped filter may be significantly lower than the order of a corresponding conventional filter. Warped filter techniques in musical acoustics have been discussed in detail by Schüssel and Winkelnkemper (1970), Steiglitz (1980), and Lansky and Steiglitz (1981), and implementation techniques for recursive direct-form and lattice filters by Härmä (2000). Also warped filter design for the body modeling in string instrument sound synthesis has been investigated (Karjalainen 1996). The time-frequency resolution of human auditory system has been considered by Penttinen (2001), and application with only one driving parameter changing the perceived size of instrument body by Penttinen *et al.* (2000). Radiated sound has also been approximated by using a digital filter processed bridge pickup signal for the guitar (Karjalainen 2000).

Virtual violins have been realized by measuring their impulse responses using, for instance, Maximum Length Sequence (MLS) technique (Farina *et al.* 1995). The impulse response is used as a numerical filter to reproduce the violin sound convolving the impulse with excitation signal.

3.2.3 Simulating other parts of the violin

Most previous efforts to integrate the violin with computing technology have focused on pitch detection, which, by itself, is not dependent on the physical interface. The bow has been used occasionally as a data input device (Trueman and Cook 1999), as has the fingerboard (Paradiso and Gershenfeld 1997).

One of the crucial deficiencies has been the lack of a decent model for the finite width of the bow that has a major role in the production of realistic sound. Paper VI introduces a method for modeling of excitation with finite width bow. According to the studies mentioned in section 3.1.4, the inclusion of torsional vibrations with correct parameter values (e.g. wave speed, impedance and damping) to a simulation model results in a more natural vibration (VI).

3.3 Expressiveness and control of violin model

In general, various control methods for synthetic instruments have been developed during the past few decades. Typically, both off-line and real-time control interfaces have been employed. Automatic performance has been studied in Friberg (1991). Interfaces for the guitar, for instance, have been written using a score based programming with *Common Lisp / CLOS* and in application to MIDI (Karjalainen *et al.* 1993). Later, the guitar models have been controlled more expressively by using tools by Laurson (Laurson 1996, Laurson *et al.* 2001, Laurson and Kuusankare 2001), such as *PatchWork*, *PWSynth*, and *Expressive Notation Package* (ENP). Such visual programming tools can be used to control also a larger variety of synthetic instruments, for example the clavichord (Välimäki *et al.* 2000), and the renaissance *ud* and *lute* (Erkut *et al.* 2001). Earlier low-level rule-based systems have been the *Chant* system (Rodet *et al.* 1984), containing a compositional environment called *FORMES* (Rodet and Cointe 1984), and a software synthesizer. This system has been designed to convincingly simulate the human voice.

Trueman (1999) has written about controlling of violin as follows:

> *"In this context, the future of the violin – be it Schubert's violin, an electric violin, or BoSSA – is seemingly rich with potential, just as its past has been rich with success. As a tool for engaging the body and mind in the processes of music making, expression, and communication, the violin is unique yet multitudinous. If anything has been learned from this exploration, it is that as violinists and composers, we must reinvent this tool to suit our needs. And as we reinvent the violin, we reinvent ourselves as well."*

In designing such instruments as presented in this thesis, we have an opportunity to create new vehicles for human expression.

The performance of music has been extensively studied by Gabrielsson (1999). This information may be used in addition to studies dealing with the control of musical instrument models. The movements of a performing player must be associated with the expression parameters. Control of the instrument involves several parallel processes whose synchrony varies depending on the musical texture. In the note list style of control, only separate and isolate notes can be sounded and slurs are approximated by keyboard style note overlap (Chafe 1989). In the study by Jensen (1996) the control of the violin consist of a set of continuous control parameters necessary to fully exploit this instrument has been proposed. The intent has been to estimate the number of parameters necessary to control a good instrument, rather than create a protocol for controlling a synthetic musical instrument.

Score-based control of simulated physical performance gestures for the physical model of bowed string instrument has been demonstrated by Chafe (1989), which is, however, considerably slower than real time. In the system the values of expressive parameters have been generated by using measurements of violin bowing gestures and distributed to the model as with MIDI control data. In the measurements the envelope curves of the magnitudes of each measured parameter have been obtained. The parameters consist of bow speed, bow pressure, and pitch. Measurements on violin bow motion and bow force have been made by Askenfelt (1986) and from the viewpoint of articulation by Schelleng (1973). The expressive synthesis model for bowed string instruments has been introduced by Takala et al. (2000).

In the paper II by the author the expressiveness of violin model synthesis has been controlled in a similar manner as in the study by Chafe (1989) and by Takala et al. (2000). The main focus in the author’s study is the expressiveness in performing multiple-stop fingerings and the related bowing.

4 HUMAN-COMPUTER INTERACTION AND THE VIRTUAL VIOLIN

A model of musical instruments should fulfill two fundamental principles: good sound quality and easy control of the important expression attributes. Both aspects have been discussed in the previous sections. Now these ideas are carried further, towards interaction between the model and a performer.

Traditional musical instruments provide compelling metaphors for human-computer interaction, both in terms of input (physical, gestural performance activities) and output (sound diffusion). Violin, one of the most refined and expressive of traditional musical instruments, combines a peculiar physical interface with a rich acoustic diffuser.

Performer of a musical instrument manipulates the performance parameter dimensions. This is embodied in physical actions, such as bowing a string. In real instruments these gestures are determined by the sound-producing mechanisms of the instrument. However, any movement producing sound carries the characteristics that gave rise to the sound (Cadoz 1988). Among sound synthesis techniques, physical modeling (see section 2.3) has the ability to decouple the synthesis of an instrument's sound from the physics of the instrument's sound-producing mechanism.

It is a well known fact, that most controllers have been based on the piano keyboard and on the assumption that each note is an isolated event with controls for its pitch, duration, timbre, and amplitude that can not interact (see e.g. Roads 1996). Investigations by Friberg *et al.* (Friberg 1991, Friberg *et al.* 1991) have presented rules for improvement of computer performance introducing the most relevant parameters of expression. For most musical instruments, a “one-to-one” mapping is the exception rather than the rule (O’modhrain 2000). Even though feedback of gestures is an important issue in controlling a virtual music device, the key issues in this thesis are the auditory feedback and the musical knowledge in instrument performance (V).

The importance of an asynchronous hierarchical structure with the performance gesture instead of the score-based approach in communication with synthesis algorithms has been brought forward by Cadoz (1988), Chafe

(1989), and McMillen (1994), since players of most of the musical instruments are in direct contact with the vibrating elements. Mechano-receptors sensitivity of human skin in musical instrument vibration production has been investigated to be within the same frequency and amplitude range (Askenfelt and Jansson 1992).

A hierarchical structure that can simultaneously represent both low-level movement trajectories and higher-level musical articulation trajectories provides an ideal model on which to build control architecture for gesture-driven interaction (O'modhrain 2000).

Several applications in virtual environments and musical instruments as well as controlling devices, such as panels, pads, and wearable devices have been created and investigated. Discussion of these in more detail is, however, out of the scope of this thesis, since the focus is placed on the bowed strings. A good overview on the historical perspectives of the topic has been given by Winkler (1995).

Violinists often talk about the vibrations they receive from the violin into the jaw and head. This connection, vital for many players, is known as *haptic*[10] connection. In this study the haptics is considered from the viewpoint of using gestural controlled virtual technologies and of hearing and feeling the sound in space (V). An introduction to the history of haptics in music has been given by Gillespie (1999). The combination of haptics and simulation in application to improved performance in human-computer interaction and skill transfer has been extensively studied by O'modhrain (2001).

The violin is an instrument, which shares the control characteristics of most of the other instruments in the family of bowed strings. The control of the violin has been investigated in various studies (e.g. Jensen 1996, Jehan 2001, Jehan and Schoner 2001, Goudeseune 1999), and a set of continuous control parameters necessary to fully exploit this instrument has been proposed by Jensen (1996). The intent in Jensen's studies has been to estimate the number of parameters necessary to control a good instrument, rather than to create a protocol for controlling a synthetic musical instrument. Jehan (2001) used the violin hardware of Jensen in his study, where the computed signal from controlling device is synthesized perceptually. The technique is based on modeling and controlling timbre. Goudeseune (1999) uses a *SpacePad* motion tracker to measure the violin position and orientation and magnetic sensors for the relative position of the bow and the violin, which are further used for spatialization of the sound output. Trueman and Cook (1999) have built an instrument that includes elements of both the violin's physical performance interface and its spatial filtering audio diffuser. The system includes a sensor-bow, a sensor-fingerboard, an array of bowed sensor-sponges, and a 12-channel spherical speaker array. Violin control parameters, such as non-discrete left-

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[10] The term haptic is derived from the Greek word haptesta (to touch), and it refers to combined feedback from tactile sensors in the skin and kinesthetic sensors in muscles and joints.

hand pitch, right-hand bow velocity, force, and orientation controls are applied in physical violin synthesis model by the author in papers II and V.

In the late 1980s, Jon Rose investigated a sensor-bow. The system consists of a pressure sensor under the index finger and a sonar sensor to detect bow position, which is a similar approach to that of Trueman's (1999). Chafe has used a cello bow with a bend sensor in the middle of the bow and an accelerometer in the frog. The bend sensor can correspond to bow pressure, or to any other action that causes the bow to bend. The accelerometer is a microscopic weight on a spring that can function as a position sensor or a motion sensor. In the work of Chafe the bow acts as an instrument, not as a measurement device. The R-Bow by Trueman and Cook is similar to Chafe's, except that they have mounted two *force-sensing resistors* (*FSR*) between the stick and hair on light foam (Trueman 1999). The same study introduces the *Fangerbored*, which is a violin fingerboard with a linear position sensor that can be played with traditional left-hand technique but on a single string. Furthermore, the *Bonge*, motivated by the bowed string, consists of four bowed "sponges". Each "string" consists of a small piece of foam-covered wood resting freely between two fixed FSRs. Each string can be bowed, creating a data point that depends on direction, pressure, and bow speed. Negative values indicate up-bow, and positive values indicate down-bow. A microcontroller converts the data to MIDI continuous control messages (Trueman 1999, Trueman and Cook 1999).

The *SuperPolm* violin controller has been developed by Goto, Pierrot and Terrier (Wanderley *et al.* 2001). Seven sensors integrated in a violin-like controller provide several continuous output variables. The ensemble provides an interface that is able to control different synthesis parameters departing from a gesticulation similar to that of a violin player (Wanderley *et al.* 2001).

In the work of Burtner and Serafin (2000), physical models of bowed string instruments have been investigated by their extended performance techniques. The models are based on the digital waveguide techniques (DWG; see sections 2.3 and 3.2) and they are applied using *Max/MSP* graphical programming interface (Zicarelli 1998) and controlled with *Metasaxophone* (Burtner and Serafin 2000). Expressive controllers for physical models of bowed strings have been further discussed by Serafin *et al.* (2001), where the controlling scheme is extended by using devices such as graphic tablet and *vBow* (Nichols 2000) hardware.

Machover, Gershenfeld, and Paradiso have constructed a set of musical instruments that they call *hyperinstruments*. Concerning the violin family, they built a variety of sensors and mounted them on the instrument in an effort to measure performance gestures (Paradiso and Gershenfeld 1997). The *hypercello*, one of the introduced hyperinstruments, consists of a sensor-bow that can transmit information regarding finger pressure, as well as wrist and bow position, and a set of sensors that detect left-hand fingerboard position. This data is sent to a network of computers for analysis and control of synthesis, samplers, and signal processors.

By Schoner *et al.* (1998) the sound output of a violin has been modeled from the gesture data captured on a muted instrument. The system is based on neural network that has been trained to learn the mapping from physical gesture input to audio parameters. During synthesis the network generates an audio output. The gesture input has been measured with a complex sensing hardware setup. Schoner *et al.* (1998) used *Cluster-Weighted Modeling* to predict a spectral sound representation giving physical input to the instrument.

The significance of haptic feedback for virtual musical instruments has been investigated by O'modhrain (2000). It was noticed that the haptic feedback could improve a player's ability to learn the behavior of a virtual musical instrument.

In the virtual violin control system by the author (V) the left-hand and right-hand gestures are merged in a way that permits their effects to interact in the synthesis. Gesture segments can coordinate changes in several parameters at once, such as the bowing location and left-hand position, as well as track evolving phrase controls.

5 SUMMARY AND CONCLUSION

5.1 Contributions to modeling and model-based sound synthesis

This thesis comprises a thorough overview in violin modeling. Various parts of this acoustically complicated system are discussed in detail. The development of new techniques and results described in this thesis is motivated by practical needs for a physical model of violin that is more reliable and more realistic physical model of a violin than what has been developed to date.

The aim of this thesis was to develop an improved violin physical model using the digital filtering tools of DWG (Digital Waveguide) modeling (e.g. Smith 1992). Though many crucial aspects in instrument modeling are discussed at least in moderate level in the thesis, the essential part of it deals with the commonly acknowledged problem of modeling the finite-width excitation in the bow-string interaction (publications III and VI). This thesis introduces a novel way to model the finite-width excitation, referred to as the FVEP (Fractional Variable Excitation Point) model. The model is based on the application of the fractional delay (FD) filtering methods (e.g. Välimäki 1995). The model is carried out to achieve an improved quality of sound synthesis of the violin.

Measurements and analyses of real violin sounds have been utilized to obtain improved performance in violin physical modeling. In particular, investigation of both the instrument body and the strings has been used to obtain a parameterization for digital filtering in separate body resonance mode modification with various filter designs (I), as well as a new solution for frequency-dependent damping at varying finger position (IV).

The expressive musical processes involved in playing a violin physical model are discussed in publication II, especially from the viewpoint of the constraints of left-hand fingering. The violin controlling paradigm is accomplished for four-string coupling violin model using control parameters such as bow velocity and pressure.

In publication V, the violin model is further applied in a virtual reality environment. This publication introduces a new human-computer interaction scheme, where a six-degrees of freedom commercial motion tracking device is used to control the physical violin model by the movements and gestures of the player, thus producing the synthetic sound of the violin in real-time.

5.2 Main results and scientific contribution of the thesis

This chapter summarizes the main results of the publications.

The violin body mode analysis, modeling, and modification are carried out in publication I. Two bodies of violins of different quality are compared and the most prominent modes of vibration are modeled for sound synthesis. The body model synthesis is carried out by using both commuted and physical modeling synthesis techniques, and body modes are also implemented at various sampling rates, thus using a multi-rate implementation. This publication is the first to apply ILFR's (Interpolated Low-Frequency Resonator) in violin sound synthesis.

Publication II discusses aspects of expressiveness in playing the physical model of violin. The emphasis is on different left-hand fingering sets that can be used when playing a complete four-stringed violin physical model with coupled strings.

Each string model was excited simultaneously during the play with different bowing styles by means of controlled bow force and velocity value distributions. To the author's knowledge, article II is one of the few studies dealing with the expressiveness and bowing styles with a four-stringed physical violin model. The effect of reducing the number of string models in the violin model on the synthetic sound quality in playing of chords was investigated. Also the limitations of a real violin compared to a modeled one were discussed. The sound syntheses with both the complete and the reduced models were produced and presented.

Publications III and VI comprise the most essential part of this thesis; they discuss the problem of finite-width bowing in bow-string interaction and provide a novel method for modeling this phenomenon. This FVEP (Fractional Variable Excitation Point) model is based on fractional delay (FD) filtering methods. The computationally economic new method provides an improved sound quality in physical modeling synthesis. Publication III introduces the idea, and publication VI provides a detailed description of the model. In publication III, the first proposal for modeling the finite width bowing excitation with fractional delay techniques is given, and the general idea of the modeling procedure as well as a few experiments are presented. In publication VI, a detailed description of the new method for modeling finite width bowing, referred to as Fractional Variable Excitation Point (FVEP) method, is discussed both theoretically and experimentally. The general mathematical formulation of the model is provided and simulations carried out with the model are described. The FVEP method is also applied to other modeled properties of violin string, such as torsional vibrations.

Publication IV presents an analysis, estimation, and synthesis of the frequency dependent damping at the fingering point. The results obtained using the analysis and optimization tools are employed for designing a digital filter to be used with the physical model of violin. As a result, a parameter dependency of damping of the sound harmonics over the length of each string of the violin is obtained. This provides an improved model of the damping with various fingerings.

In publication V, the violin physical model is applied in a virtual environment. A six-degrees of freedom commercial motion tracking device is used to control the actual physical violin model by the movements and gestures of the player, thus producing synthetic sound of violin in real time. The technical details of the system are presented and the advantages and limitations are discussed. To the author's knowledge, this publication is the first to introduce this kind of a real-time tracker controlled violin.

5.3 Contribution of the author

In publication I, the author carried out the measurements and analyzed the measurement data, as well as implemented the filtering applications. The article was written and the companion web-page was prepared by the author. Prof. Vesa Välimäki contributed to this article in collaborating the writing of the paper and advising in filter implementation.

The author was responsible for the research in publication II.

The author was responsible for the research in publication III.

The author was responsible for the research in publication IV.

The author was responsible for the research in publication V.

The author was responsible for the research in publication VI. The publication was contributed with collaboration of Prof. Petri Toiviainen with some advice on the mathematical formulation and writing.

5.4 Discussion and future directions

The violin is easily considered as one single type of an instrument. As discussed in section 3.1 a distinction between instruments such as Baroque, Classical, and Renaissance can be clearly made. A good addition to traditional modeling of bowed strings would be possible by taking advantage of the properties of these different styles of the instrument in application to physical modeling.

The admittance of the bridge-body system needs to be extensively investigated. A major role in couplings of the instrument is dealt with the admittance, mainly via the bridge. The virtual admittance for modeling purposes for the guitar was discussed carefully in (Nackaerts et al. 2002).

Furthermore, more refined paradigms in body modeling may start taking place when the computing power of even regular home computers has attained a certain level so that more and more complicated systems can be computed in real time. Certain models, still too complicated for real-time use, are developed, for instance, by Derveaux *et al.* (2002) and Chaigne (2002) for sound radiation of the guitar body. The models utilize a mixture of a few modeling techniques (time-domain simulation, FEM, PDE) to get the whole body geometry interact with the air inside the cavity and surround.

As a continuum to the investigations of Woodhouse (1992), the importance of time-dependent rosin friction needs to be considered in more detail. Even though the behavior of the friction in rosin melting process has been quite well studied (Smith and Woodhouse 2000, Woodhouse *et al.* 2000, Guettler 2002), there is still a need for development of a feasible rosin friction model for friction driven musical instrument sound synthesis purposes.

The expressiveness and control over the models is still a vital task to be considered. In this thesis, the control of violin model is handled either in MATLAB simulations by long physical parameter value vectors, or by movements and gestures of the player in real-time virtual environment, though other needs and used interfaces in controlling such models should also be taken into account. One noteworthy tool in application to this would be the expressive notation packages, such as that of Laurson's (Laurson 1996, Laurson and Kuusankare 2001).

Even though a large number of different methods with very rich variety in parameter space have been investigated during the past decades, there is still space for new innovations in the field of bowed-string instrument sound synthesis. The bowed-strings should be investigated more accurately, for instance, by mixing different styles of sound-source object modeling with model-based sound synthesis methods by means of human auditory system that considers how the sound is perceived by a human being.

In virtual reality and multimedia applications of today and future *MPEG-4* provides a set of tools that enable construction and concise transmission of audiovisual scenes as composites of several components of content such as video clips, computer graphics, recorded sound, and parametric sound synthesis (e.g. Scheirer 1998, Scheirer *et al.* 1999). *AudioBIFS*, one of the tools collected under *BIFS* (BInary Format for Scene description), is a specific subset that controls the composition of sound scenes, and uses streaming audio, interactive and terminal-adaptive presentation, three-dimensional spatialization, and dynamic download of custom signal-processing effects. Therefore, the presented virtual violin can be seen to be fully applicable within the MPEG-4, the use of which needs to be considered in the near future.

Future extensions to the human-computer interaction in virtual musical instruments need to be extensively studied. It has been investigated that computer-generated haptic feedback can improve performance of manipulation tasks in virtual or telepresence environments (O'modhrain 2001). Therefore, the gestural controlled virtual musical instruments, such as the virtual violin, and their controlling paradigm in a more cognitive manner, are with no doubt increasingly essential tasks for further examination.

YHTEENVETO

Virtuaalinen viulu digitaalisella alueella. Viulun fysikaalinen mallintaminen ja mallipohjainen äänisynteesi, sekä sen vuorovaikutteinen soveltaminen virtuaalitodellisuusympäristössä.

Tämä väitöskirja luo laajan katsauksen viulun mallintamiseen. Useita osia tästä akustisesti erittäin monipuolisesta järjestelmästä käsitellään yksityiskohtaisesti. Väitöskirjan syntymistä ja siinä esitettyjen uusien tekniikoiden kehittämistä ovat ohjanneet pitkälle käytännön syyt. Tutkimuksen tuloksena on pyritty luomaan aiempia fysikaalisia malleja luotettavampia ja todenmukaisempia malleja.

Väitöskirjan tarkoituksena oli kehittää paranneltu viulun fysikaalinen malli käyttäen digitaalisen signaalinkäsittelyn työkaluja, erityisesti digitaalisia aaltojohtomalleja. Vaikka väitöskirjassa on käsitelty useimpia merkittäviä mallinnukseen liittyviä seikkoja vähintään jollakin tasolla, on merkittävin käistellyistä aiheista yleisesti tiedostettu äärellisen jousen leveyden mallintamisen ongelma jousi-kieli –vuorovaikutuksessa (julkaisut III ja VI). Väitöskirjassa esitellään uusi, erilainen keino mallintaa äärellisen leveyden aikaansaamaa herätettä. Menetelmä on muuttuvan murtoviivepisteen malli (FVEP, Fractional Variable Excitation Point). Malli perustuu murtoviivesuodinten (FD, Fractional Delay) soveltamiseen (esim. Välimäki 1995). Tässä työssä esitelty fysikaalinen malli on kehitetty tuottamaan entistä laadukkaampaa synteettistä jousisoittimen ääntä.

Fysikaalista mallinnusta varten on mitattu ja analysoitu oikeiden viulujen ääntä. Tarkemmin sanoen työssä on tutkittu sekä viulujen kaikukopan resonointia että kielten värähtelyä. Näiden mittausten ja tulosten analyysin perusteella on saatu määritettyä esimerkiksi kaikukoppaa mallintavien suodinrakenteiden parametrejä (I). Lisäksi tutkimuksessa on esitelty uudenlainen ratkaisumalli taajuusriippuvaiselle vaimenemiselle eri sormitusvaihtoehdoissa (IV).

Viulun fysikaaliseen malliin ja sillä soittamiseen liittyvää ekspressiivisyyttä erityisesti vasemman käden sormitusten näkökulmasta on käsitelty julkaisussa II Viulumallin kontrollointia on tarkasteltu realistisen kaltaisen neljä kielimallia käsittävän kytkeytyneen fysikaalisen mallin avulla syöttämällä ohjausparametreinä mm. jousen nopeus- ja painetietoa.

Julkaisussa V viulumallia on edelleen sovellettu keinotodellisuusympäristöön. Julkaisussa esitellään uudenlainen ihmisen ja tietokoneen välisen vuorovaikutuksen malli, jossa kuuden vapausasteen liikkeenkaappausjärjestelmää sovelletaan ohjaamaan fysikaalista viulumallia soittajan liikkeiden avulla ja tuottamaan synteettistä, reaaliaikaista viulun ääntä.

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