UNIVERSITI PUTRA MALAYSIA

TIME-VARYING SPECTRAL MODELLING O F THE SOLO VIOLIN TONE

ONG BEE SUAN

FEM 2000 8

TIME-VARYING SPECTRAL MODELLING OF THE SOLO VIOLIN TONE

By

ONG BEE SUAN

Thesis Submitted in Fulfilment of the Requirements for the Degree of Master of Science in the Faculty of Human Ecology Universiti Putra Malaysia

April 2000

Abstract of thesis presented to the Senate of Universiti Putra Malaysia in fulfilment of the requirements for the degree of Master of Science.

TIME-VARYING SPECTRAL MODELLING OF THE SOLO VIOLIN TONE

By

ONG BEE SUAN

April 2000

Chairman: **Minni Ang Kim Huai, Ph.D.**

Faculty: **Human Ecology**

The analysis of the spectrum of a single violin tone, to better understand how the various partial components contribute to the sound produced, is undertaken. The analysis involves determining which partials are present and how these partials evolve with respect to time. The short-time Fourier transform is used to implement a solution for the time varying spectra by chopping the sound into short segments called windows and analysing each segment sequentially. The MATLAB digital signal processing software was used in both the analysis and resynthesis stages of this research. Parameters extracted through analysis are used for resynthesis purposes. Results indicate that spectrum changes over time contribute significantly to the timbre of the

violin tone. A slight shifting of the fundamental frequency was also observed in the sound spectrum of all the sub-sections of the waveform, although this shifting was most marked in the attack and release portions of the ADSR envelope. The results also showed that the intensity of the fundamental harmonic was weaker in the initial attack stage, only dominating when the timbre of the tone stabilised. Within the release portion, inharmonic overtones were shown to occur in the upper partials of the sound spectrum. Finally, the resynthesis process reduces the required hard disk capacity by about 93.8 percent compared with the sampled waveform, while at the same time producing an audible tone almost indistinguishable from the original.

Abstrak tesis yang dikemukakan kepada Senat Universiti Putra Malaysia sebagai memenuhi keperluan untuk ijazah Master Sains.

MODEL SPEKTRUM BERSANDARKAN MASAUNTUK NADA BIOLA SOLO

Oleh

ONG BEE SUAN

April 2000

Pengerusi: **Minni Ang Kim Huai, Ph.D.**

Fakulti: **Ekologi Manusia**

Spektrum nada biola solo dianalisis untuk memahami bagaimana pelbagai komponen separa menyumbang kepada bunyi yang dihasilkan. Prosedur analisis ini merangkumi pengenalpastian komponen separa yang hadir di dalam spektrum serta bagaimana komponen separa ini berubah dengan masa. Jelmaan Fourier masa-singkat digunakan untuk mengimplementasi penyelesaian bagi spektrum yang bersandarkan masa secara membahagikan isyarat bunyi kepada segmen kecil yang dipanggil tetingkap lalu menganalisa setiap segmen mengikut turutannya. Perisian pemerosesan isyarat digital MATLAB digunakan dalam peringkat analisis serta peringkat sintesis semula. Parameter yang diperolehi melalui proses analisis digunakan untuk tujuan mensintesis semula nada biola solo tersebut. Keputusan yang diperolehi menunjukkan bahawa perubahan spektrum terhadap masa

mempunyai kesan signifikan ke atas timbre nada viola. Satu cerapan lain yang diperolehi melibatkan anjakan kecil frekuensi asasi dalam semua sub-bahagian gelombang spektrum bunyi yang dikaji. Anjakan paling nyata diperolehi dalam bahagian permulaan dan bahagian akhiran sampul ADSR. Keputusan juga menunjukkan bahawa keamatan harmonik asasi adalah lemah di bahagian awal peringkat permulaan bunyi dan hanya dominan setelah timbre nada menjadi stabil. Dalam bahagian akhiran, komponen separa yang bukan harmonik didapati berlaku dalam separa atasan spektrum bunyi. Proses sintesis semula pula didapati menghasilkan penjimatan keperluan simpanan data sejumlah 93.8 peratus berbanding dengan gelombang sampel asli, disamping menghasilkan nada kedengaran yang hampir tidak dapat dibezakan daripada nada asal.

ACKNOWLEDGEMENTS

I would like to thank my principal supervisor, Dr Minni Ang Kim Huai, for her patient and detailed guidance throughout the course of my study and for providing me with all the materials and facilities which I needed for this research such as books, journals and laboratory facilities. I appreciate the opportunity provided to me to present my research paper at the Malaysian Science and Technology Congress, MSTC'99, Symposium C organised by the Confederation of Science and Technology Associations in Malaysia (COSTAM) in Johor Bharu, which enable me to have my first publication. I would like to also thank my second supervisor, Dr Veeraraghavan Prakash, for his helpful tips especially with regards to the section on the Fourier analysis. My thanks also to the other members of my supervisory committee, Ms Martha Lee Jin Ai and Mr Rick Craig Shriver, for their encouragement and input. Thanks also to Dr Robert Tee for his input. I also would like to thank the UPM Graduate School for supporting my study through the Pasca Siswazah scheme. I appreciate my research colleagues, Yaw Feng, Julie and Yoke Fun, and also Angelin and Auntie Rano for providing a cheerful atmosphere in our graduate lab, which helped me to relax and enjoy my work. Finally, I would like to thank my mom and my dad for their support, understanding and love.

I certify that an Examination Committee met on 3 April 2000 to conduct the final examination of Graduate Student on her Master of Science thesis entitled "time-varying spectral modelling of the solo violin tone" in accordance with Universiti Putra Malaysia (Higher Degree) Act 1980 and Universiti Putra Malaysia (Higher Degree) Regulation 1981. The Committee recommends that the candidate be awarded the relevant degree. Members of the Examination Committee are as follows:

Dr. Rozumah Baharudin, Ph.D
Faculty of Human Ecology
Universiti Putra Malaysia
(Chairman)

Dr. Minni Ang Kim Huai, Ph.D
Faculty of Human Ecology
Universiti Putra Malaysia
(Member)

Dr. Veeraraghavan Prakash, Ph.D
Faculty of Engineering
Universiti Putra Malaysia
(Member)

Martha Lee Jin Ai, M.A
Faculty of Human Ecology
Universiti Putra Malaysia
(Member)

Rick Craig Shriver, M.A.
College of Communications
Ohio University, USA
(Member)

MOHD. GHAZALI MOHAYIDIN, Ph.D,
Professor/Deputy Dean of Graduate School,
Universiti Putra Malaysia

Date: 0 5 APR 2000

This thesis submitted to the Senate of Universiti Putra Malaysia and was accepted as fulfilment of the requirements for the degree of Master of Science.

KAMIS AWANG, Ph.D,
Associate Professor
Dean of Graduate School,
Universiti Putra Malaysia

Date: 11 MAY 2000

DECLARATION

I hereby declare that the thesis is based on my original work except for quotations and citations which have been duly acknowledged. I also declare that it has not been previously or concurrently submitted for any other degree at UPM or other institutions.

signed

Candidate
Ong Bee Suan
Date: 4/4/2000

TABLE OF CONTENTS

LIST OF TABLES

LIST OF FIGURES

LIST OF ABBREVIATIONS

STFT	short-time Fourier transform
DTFT	discrete-time Fourier transform
DFT	discrete Fourier Transform
FFT	fast Fourier transform
FT	Fourier transform
VCO	voltage controlled oscillator
MIDI	musical instrument digital interface
kHz	kilohertz
PSD	power spectral density
dB	decibel
Hz	hertz

CHAPTER ONE

INTRODUCTION

Music synthesis is a method or technique by which a fundamental waveform is generated using a voltage or computer controlled oscillator, with changes being made to that waveform to yield different sounds by using different techniques, such as modulation and filtering. Early pioneers of music synthesis include John Chowning and Max Mathews in the 1960's. During recent years, music synthesis technology has developed very rapidly and many different new music synthesis techniques have been developed: for example, Physical Modelling (Smith, 1992; Lehman, 1996) and Spectral Modelling (Serra & Smith, 1990). Recently, music synthesis techniques have become more and more intricate and sophisticated. Some of them even use a combination of hardware and software to generate new sounds. Many imitation sounds of various musical instruments and sound effects have been created and used in synthesisers and electronic keyboards. In music synthesis, many different techniques are used to generate sound. According to Miranda (1998), the basic classes of synthesis techniques are Loose Modelling, Physical Modelling, Spectral Modelling, and Time Modelling. Although different techniques can produce the same sound, for example the sound of a violin, different techniques may produce a different quality of sound. Due to the demand for better quality and for new and interesting sounds from customers, commercial profit-motivated organisations invest a substantial amount of funding to encourage research in this area. The aim of this research includes the invention of new synthesis techniques to produce a better quality of synthesised sound, to improve existing synthesis methods, and to reduce storage requirements. This has led to new synthesis techniques, such as the Karplus-Strong Technique invented by Karplus and Strong in 1983. According to Karplus and Strong (1983), although this synthesis technique lacks of versatility compared with other synthesis techniques such as additive synthesis, it is inexpensive enough to be implemented on microprocessors and provides surprisingly rich timbres. In addition, analysis of the sound spectrum of musical instruments has evolved from the traditional analysis method [Fourier transform] to the Fast Fourier transform (FFT) method, which increases the speed of analysis. Much research also has been done on data reduction, which lessens storage problems and accelerates the process of synthesis. These include Grey and Moore (1977), Charbonneau (1981), Sandell and Martens (1995) and recently McAdam *et al.* (1999). The development of research on the analysis of musical instruments has also grown rapidly, including an ever increasing range of musical instruments such as the piano and harpsichord (Weyer, 1976), plucked-strings (Karplus and Strong, 1983), double bass (Abbas, 1989), winds (Keefe, 1992), violins (Miller, 1993) and others.

Statement of the Problem

The spectral model of the solo violin tone is constructed from the intensities of the various partial components of its waveform, and these intensities need to be obtained experimentally. A complete mathematical model of a musical tone consists of periodic and non-periodic functions. Periodic functions are generally modelled as summations of simple sinusoids, according to Fourier's theorem. Non-periodic functions, such as the amplitude envelope, transient sounds and residual noise, contribute towards the realism of the tone, but are not part of the model of the present research, which is limited to the modelling of the periodic functions only.

The periodic function is contributed by the addition of the partials that occur in the sound waveform. This can be shown through the analysis of the sound waveform using spectrum analysis to identify the partials that occur. The sound waveform can be defined as in the formula below:

$$y = a_1\sin(2\pi f_1 t_1) + a_2\sin(2\pi f_2 t_1) + a_3\sin(2\pi f_3 t_1) + \ldots + a_n\sin(2\pi f_n t_1)$$

Equation (1)

where,

y = the waveform of the sound signal
a = the harmonic relative amplitudes for the sound
f = frequency of the harmonics that occurs in the sound signal
t = the time at which the waveform is captured

The Fourier series equation usually assumes that the waveform does not change over time. However, according to the equation above, the waveform does change with time. Therefore, the exact contribution of each of these modes is time-varying with respect to the overall sound. This exact contribution needs to be determined through research.

Objective of the Study

The objective of the study is to analyse the spectrum of a single violin tone in order to better understand how the various harmonic or partial components contribute to the sound produced. The analysis involves determining which partials are present and how these partials evolve with respect to time. The parameters obtained in this way may then be used to create a mathematical model of the violin tone consisting of periodic functions only, that is the spectral model. This spectral model may be used to resynthesise the violin tone, creating a more compact way of playing back the violin sound with respect to data storage requirements.

Significance of the Study

The significance of the study is to obtain a better model of the violin tone in order to provide a better understanding to the violin sound. Through the understanding of the spectrum of the instrument sound, one can control the parameters of the sound and create a better or higher quality sound for synthesisers, sound card, sound modules and others for commercial purposes. On the other hand, the spectral parameters can also be manipulated to create new sounds that can be used for compositional purposes.

Design of the Study

The research is conducted in four main sections: the testing of the methodology, the recording of the violin tone, the analysis of the recorded data, and the resynthesis using parameters obtained from the analysis.

The methodology is tested through the analysis of a digitally generated tone [using a mathematical equation]. After that, the Fast Fourier Transform is applied to the tone to analyse the spectrum of the sound signal. Finally, results obtained from the analysis are compared with the original mathematical equation to ensure that the analysis-resynthesis methodology is reliable and can be used in this research.

The violin sound is recorded using a digital hardisk recorder. After that the sound signal is transferred to the computer for noise reduction and analysis purposes. In the third section, the data analysis process is carried out. The sound signal is divided into four portions [attack, decay, sustain and release] and each of these portions is further subdivided into even smaller portions. After that, the Fast Fourier Transform is applied to all these small portions in order to obtain the spectrum changes of the sound signal. Finally, the result of the analysis is used for resynthesis purposes and compared with the original acoustic musical instrument sound.

Organisation of the Thesis

In Chapter Two, the literature review is undertaken. This focuses on definitions of synthesis and related terms, the history and development of music synthesis technology and current research trends.

In Chapter Three, the methodology of the whole research is discussed in precise detail. This chapter contains four main sections that explain the methodology of this research. It includes some explanations of the Fast Fourier Transform, type of window, sampling rates and others parameters which are used in this research. Besides that some rules about the analysis process are discussed.

Chapter Four contains the results and discussion related to the research. In this chapter, some tables and some three-dimensional graphs are plotted.

Chapter Five contains conclusions of the study. This chapter ends with suggestions for further study. The literature review is now considered further.

CHAPTER TWO

LITERATURE REVIEW

This chapter contains the review of related literature. It begins with a description of digital music synthesis. The spectral modelling synthesis technique, which is used in the present research, is then documented. In addition, the significance of music sound characteristics with respect to timbre is explained. This is followed by an account of the significance of spectral and temporal parameters. Current developments and research on musical sound characteristics and spectrotemporal parameters are highlighted. Spectral analysis [which is one of the spectral modelling synthesis techniques used in this research] and its analysis methods such as the Short-time Fourier transform and the Fast Fourier transform are also explained. Finally, the theoretical basis for the resynthesis process is described.

Digital music synthesis

Digital sound synthesis generates a stream of numbers representing the samples of a sound signal waveform. There are many different general methods created for the purpose of digitally synthesising musical instrument sounds. These wide ranges of synthesis techniques have been described in the literature (Moore 1977; De Poli 1983; Gordon 1985; Road 1996). Different techniques focus on different aspects. Some of these techniques have been built by emulating the mechanics of a natural sound production process. This approach is known as physical modelling (Smith, 1992; Lehman, 1996). Other techniques use samples, which are short recorded segments of sounds produced by real acoustic sound sources such as musical instruments (Russ, 1996). Yet other techniques analyse the sound spectrum of a real instrument in order to obtain musical sound characteristics for resynthesis (Serra & Smith, 1990).

Spectral Modelling Synthesis

Spectral modelling synthesis was developed by Xavier Serra and Julius Smith, in 1990 (Vaggione, 1996). The spectral modelling synthesis technique is a set of techniques and software implementations for the analysis, transformation and synthesis of musical sounds. The aim of this work is to get general and musically meaningful sound representations based on analysis of musical sound characteristics, from which musical parameters might be manipulated while maintaining high quality sound. (Serra, 1998). These techniques employ parameters that describe the sound spectrum, regardless of the acoustic mechanisms that may have produced them. Spectral modelling synthesis technique is developed through Fourier analysis, which considers a pitched sound to be made up of various sinusoidal components, where the frequencies of the higher components are integral multiples of the frequency of the lowest component. (Miranda, 1998).

Spectral modelling techniques can be used for synthesis, processing and coding applications, while some of the intermediate results might also be applied to other music related problems, such as sound source separation, musical acoustics, music perception, or performance analysis (Serra, 1998). However, the great advantage of these techniques compared with plain sampling is that musicians can manipulate these coefficients in a variety of ways, in order to create new sounds such as sound morphing, which can be achieved by varying the coefficient accordingly. (Miranda, 1998).

The Significance of Music Sound Characteristics

As mentioned earlier, spectral modelling techniques employ parameters, which describe the full representation of the physical properties and the behaviour of the sound signal, thus musical sound characteristics are very important.

Many studies have been done on musical sound characteristics, especially from the perspective of sound timbre and sound identification. Acoustic characteristics which correspond with physical and behavioural properties of sound sources [such as spectral centroid and inharmonicity] are most important for discrimination of different sounds (Martin, 1998). Through the use of these characteristics, an idea of creating a computer system which can recognise sound sources in a complex environment with the use of computational auditory scene analysis [also called CASA] has been proposed (Martin, 1998).

McAdam *et al.* (1999) did research on the discrimination of musical instrument sounds resynthesized with simplified parameters [for the purpose of data