TIME-VARYING SPECTRAL MODELLING OF THE SOLO VIOLIN TONE

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ABSTRACT

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 slicing the sound into short segments called windows and analysing each segment sequentially. A 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.

Kata kunci: Teknik sintesis muzik; penjanaan nada digital; muzik komputer; muzik digital; permodelan spektrum; akustik biola.

ABSTRAK

Analisis spektrum nada tunggal biola solo teleh dijalankan untuk memahami kesan pelbagai komponen gelombang separa ke atas bunyi nada biola. Analisis ini melibatkan pengenalpastian komponen separa yang wujud dalam gelombang bunyi nada biola solo, serta bagaimana komponen ini berubah dengan masa. Transformasi Fourier tempoh-pendek telah digunakan untuk mengimplementasi penyelesaian bagi spectrum yang berubah dengan masa secara membahagi bunyi kepada segmen pendek yang digelar tetingkap dan seterusnya menganalisa setiap segmen secara berturut-turut. Perisian memproses signal secara digital telah digunakan dalam fasa analisis dan fasa sintesis semula. Parameter yang diperolehi melalui analisis digunakan untuk tujuan mensintesis semula gelombang bunyi tersebut. Data menunjukkan bahawa perubahan spectrum dengan masa banyak mempengaruhi timbre bunyi nada biola. Frekuensi asas dicerap beralih sedikit dalam semua bahagian kecil bentuk gelombang, dengan peralihan yang paling nyata di bahagian permulaan dan akhir sampul ADSR. Keputusan juga menujukkan bahawa harmonic asas lebih lemah di awal bahagian permulaan gelombang, dan hanya menjadi kuat setelah timbre nada menjadi stabil. Komponen separa inharmonik diperhatikan wujud dalam bahagian akhir bentuk gelombang, dalam julat separa tinggi spectrum bunyi. Proses sintesis semula menjimatkan 93.8 peratus ruang cakera liat berbanding dengan rakaman gelombang asal dan menghasilkan kualiti bunyi yang hampir sama dengan yang asli.

INTRODUCTION

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.

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 Equation 1 below:

$$y = \sum_{n=1}^{N} \sum_{m=1}^{M} a_n \sin(2\pi f_n t_m + \varnothing_n) \qquad [1]$$

where,

y = the waveform of the sound signal

a_n = the relative amplitude of the nth harmonic

f_n = frequency of the nth harmonic

t_m = the time at which the waveform is captured

$\varnothing_n$ = the phase of the nth harmonic

The Fourier Series equation usually assumes that the waveform does not change over time. However, according to the formula 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.

The objective of the study is to analyse the spectrum of a single violin tone, 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 for resynthesising the violin tone. The purpose of this resynthesis is to obtain savings in data bandwidth – in other words, to create the same sound [or as close to the original sound as possible], using minimal memory requirements.

LITERATURE REVIEW

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. 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). 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). Conclusions drawn from McAdam *et al* (1999) indicate that spectral-envelope shape [jaggedness and irregularity in the shape of the spectrum] and spectral flux [change in the shape of the spectral envelope over time] are the most important physical parameters in timbre discrimination. Literature on musical instrument sounds has suggested that there are many other acoustic features that are also useful for instrument identification such as inharmonicity (Benade, 1990), spectral centroid (Handel, 1995), and intensity (Beauchamp, 1982). Most spectral modelling techniques work in two stages: analysis and resynthesis (Miranda, 1998). Analysis is a process of digitising sound from a natural musical instrument first and then analysing it to determine the content of partial frequencies and the associated amplitude and frequency envelopes, while resynthesis is the process of synthesising sound on the basis of information derived from the analysis of another sound (Dodge and Jerse, 1997).

METHODOLOGY

Recording the Violin Tone to be used in the Analysis

The violin used in recording the musical instrument tone for this research was a *Charles Buthod* handmade French violin. The violin strings used were *Thomastik "Dominant"* violin strings, and the bow was a *Mirecort* bow. The recording of the violin sound was conducted in a quiet and acoustically dry [non-reverberant] room, with the aim of minimising reverberation and background sounds [which generate energy affecting the recorded spectrum partials]. With a sampling rate of 44100 Hz and a resolution of 16 bits, a compact disc quality violin sound was recorded. During the recording, only the monophonic [single melodic line] violin sound was recorded. This was to ensure that successful analysis could be done.

The Analysis Methodology

Spectrum analysis is important for spectral modelling, as samples alone do not inform of the spectral constituents of a sampled sound. There are two categories of spectrum analysis, which have been created to analyse the spectrum of the sounds. They are harmonic analysis and formant analysis. Harmonic analysis focuses on the identification of the frequencies and amplitudes of the spectrum components, while formant analysis uses the estimation of the overall shape of the spectrum's amplitude envelope.

The Fourier transform decomposes or separates a waveform or function into sinusoids of different frequencies, which sum to the original waveform (Hoffman, 1998). It identifies or distinguishes the different frequency sinusoids and their respective amplitudes (Brigham, 1998). The Fast Fourier Transform (FFT) is a Discrete Fourier Transform (DFT) algorithm developed by Tukey and Cooley that reduces the number of computations from the order of N_0^2 to $N_0 \log N_0$. (Hoffman, 1999). The short-time Fourier transform (STFT) implements a solution for time varying spectra by slicing the sound into short segments called windows and analysing each segment sequentially. It uses FFT to analyse these windows, and plots the analysis of the individual windows in sequence in order to trace the time evolution of the sound. The result of each window analysis is called an FFT frame. Each FFT frame contains two types of information. One is a magnitude spectrum which depicting the amplitudes of every analysed component, and the other one is the phase spectrum that shows the initial phase for every frequency component (Miranda, 1998). In this research, the phases that occur in the sound spectrum are not considered. This is due to the fact that in determining the timbre or sound quality of the wave the spectral component amplitudes are far more important than their phases (Berg and Stork, 1995).

The sample waveform of the recorded violin note is recalled and divided into four main portions, according to the attack-decay-sustain-release (ADSR) envelope consisting of the attack portion, decay portion, sustain portion, and release portion. Each portion is further subdivided into even smaller sections. The minimum duration time of each section was set at 0.1 seconds. [Portions with duration times of less than 0.1 seconds produced significant fluctuation of results when the FFT function was applied]. The sections for data analysis are also overlapped to improve the periodogram estimation of the results. Table 1 shows the details for each section of data used for the analysis.

A digital signal processing software was used in both the analysis and resynthesis stages of this research. The Welch power spectrum estimation method was used. It estimates the power spectrum by using the Fast Fourier Transform and taking the squared magnitude of the results. In order to perform the Fast Fourier Transform, parameters such as the sampling frequency, length of FFT, size of window, number of overlapping samples and type of window need to be specified. Below are the parameters that are used in this research.

Window type. Different types of windows display different types of frequency graphs. With a suitable window, the spectral leakage is diminished and the width of the spectral peaks is increased. In this research the Hanning window was used. One of the reasons the Hanning window was chosen is due to its simple implementation. The Hanning window has low side lobes, which makes it easier to pick up major frequency components. Although different functions may be used in this research, it is important to choose a suitable window function whose characteristics are best suited to the problem that is being addressed (Brigham, 1988).

Size of window. The size of the spectral window affects the variance of the estimation. The narrower the spectral window, the larger the variance of the estimation. In this research, the size of the window is set to 65536 or 2^{10} in order to produce a stable and low variance estimate.

Length of the FFT. The Length of the FFT is the number of the difference frequencies at which the power spectrum is estimated. The length of the FFT has to be a power of two, in order to employ a high-speed radix-2 fast Fourier transform algorithm. It is due to the characteristic of the radix-2 FFT routine. The Radix-2 FFT routine is optimised to perform a real FFT when the input sequence is purely real, and computes a complex FFT when the input sequence is not. This causes a real power of two FFTs to be 40 percent faster than a complex FFT, although both have the same FFT length. Thus, the length of the FFT, which is a power of two, has the fastest execution time. In order to obtain the fastest execution time, the FFT length is set to 65536 or 2^{10}. When the sequence length is not an exact power of two, an alternate algorithm finds the prime factors of the sequence length and computes the mixed-radix discrete Fourier transforms of the shorter sequences.

Sampling frequency. According to the Nyquist Theorem, to digitally encode the desired frequency bandwidth the sampling frequency must be set at least twice as high as the highest desired frequency. As a result, in order to reproduce a signal with an 11 kHz frequency range, a 22kHz sampling rate has to be selected. Otherwise aliasing [which is caused by having too few samples to show the basic waveform] will occur. Thus, a proper sampling rate will enable the same waveform to be reconstructed.

Number of overlapping samples. This refers to the number of samples in the overlapping sections. In this research, the number of overlapping samples is set to zero.

In the analysis section, there are a few rules. They are as below:

1. The size of the window must be less or equal to the length of the FFT.
2. The minimum value of the length of the FFT has to be 2^8, which is equal to 256.
3. Values for NOVERLAP, number of samples overlap and nfft, length of the FFT has to be positive integers.
4. The number of overlapping samples has to be less than the window size.

As explained earlier, the number of the FFT will determine the number of difference frequencies at which the power spectrum is estimated. The compromise between small variance and high resolution is the most important issue of the power spectrum estimation. With a small variance, a stable estimation is produced. On the other hand, with a high resolution, one can look into all the fine details in the spectrum. It is thus important to know the main purpose of the experiment and how accurate an estimation is required.

To obtain the relative amplitudes the following equation (2) was used,

$$\therefore \text{ Relative amplitude of the Harmonic, } \frac{A_n}{A_{peak}} = \text{Antilog}_{10} \frac{PSD_{relative}}{20} \qquad [2]$$

The Resynthesis Process

In order to resynthesise the original sound, the results obtained from the analysis were inserted back into the Equation 1 according to their time-varying components. All the contributing equations were concatenated, that is appended one after another according to their time-varying intervals. Through this method, the waveform produced consists of a sound spectrum that changes over time. A comparison between the synthesised sound and the original sound was then carried out. The purpose of resynthesising the sound is to find out how authentically the equation manages to generate the sound compared with the original sound. It also helps in discovering other factors needed to improve the quality of the sound produced.

RESULTS

Figure 1 below shows a screen snapshot of the recorded violin tone. It is an A4 violin tone recorded on stereo channels using a sampling rate of 44.1 kHz with a resolution of 16-bits. The complete duration of this tone is 2.575 seconds, and is stored in the `.wav` file format using 221 KB of hard disk space. This A4 violin tone, as explained previously, is divided into four portions according to ADSR parameters. These four portions are the attack, decay, sustain and release portions (Figure 1). The attack portion occurs between time T0 and T1, the decay portion between T1 and T2, the sustain portion between time T2 and T3, and the release portion between T3 and T4. Table 2 presents a summary of all these results.

Results of the Spectrum Analysis

The Attack Portion

The sound spectrum of all the four sections within the attack portion consists only of harmonic overtones [which are integer multiples of the fundamental harmonic, f]. Within the attack portion, there is an occurrence of a weaker intensity on the fundamental harmonic. In the first two sections [which are section A1 and A2], the fundamental frequency does not contain the highest power spectrum density compared with other partials in the sound spectrum. In fact, the highest power spectrum density focuses on the fifth harmonic, 5f. It is then followed by the second harmonic and then only the fundamental harmonic. The relative amplitudes of the fundamental harmonic within section A1 and A2 are only 0.7498 and 0.9469, while the second harmonic records values of 0.9710 and 0.98. The highest power spectrum density in section A1 is –82.23164 decibels which is about 2.501147 decibels louder than the fundamental harmonic, while in section A2 the highest power spectrum density is –80.958757 decibels, which is about 0.473661 decibel louder than the fundamental harmonic. Thus, one can see that the difference between the fundamental harmonic and the second harmonic in these two sections becomes smaller.

Table 3 shows the changes, with respect to time, in the relative amplitudes of the sound spectrum within the attack portion. In section A1, the relative amplitude of the fundamental frequency is only 0.7498. In section A2, the relative amplitude of the fundamental frequency starts to increase and from section A3 onwards the fundamental frequency contains the highest relative amplitude within the sound spectrum. This result indicates that the initial attack portion does not start with a strong fundamental harmonic, but is developed over time. From section A1 to A4, the relative amplitude of the fundamental frequency starts to strengthen, while the amplitude of the fifth partial starts to weaken. The sixth harmonic and the eighth harmonic are very consistent throughout. Throughout the entire attack portion, there are no changes in the relative amplitudes of the sixth and eighth harmonics.

The results also record the occurrence of a shift in frequency. In the first two sections the fundamental frequency is 442.77649 Hz, while sections A3 and A4 have a slightly different fundamental frequency. For example, fundamental frequency of A3 is 442.10358 Hz and the fundamental frequency of A4 is 441.43066. Table 4 shows the difference in the fundamental frequency of all sections within the attack portion.

The Decay Portion

Within the decay portion, the sound spectrum of all the sections consists only of harmonic partials. There are no occurrences of inharmonic partials. The sections in this portion have some differences when compared with the sections in the attack portion. In this portion, each occurrence of the fundamental harmonic always has the highest intensity [also called the power spectrum density or perceived loudness] level compared with other partials, which also occur in the sound spectrum. This is unlike the attack portion, in which there are some occurrences of the fundamental harmonic with an intensity level lower than other harmonic partial values, even though the perceived tone is still the A4 note. In the decay portion, all the peak relative amplitude values focus on the fundamental harmonic only.

A shift in the frequency is also found to occur within this portion. Table 5 displays the shift frequency of each section within the decay portion. In this portion, although there is a shift in frequency, most of the fundamental frequency focuses on 441.43066 Hz. There is only one fundamental frequency that focuses on another frequency [442.10358 Hz]. This occurs in the first section within the decay portion [also called D1].

Table 6 displays the changes over time in the relative amplitudes of the sound spectrum within the decay portion. This table clearly displays the consistency of the relative amplitudes of the eighth, tenth, twelfth and fourteenth harmonics. Throughout the decay portion, there is no change in their relative amplitudes, which remain at 0.1.

The Sustain Portion

As in the previous two portions, all the sections in the sustain portion consist of only harmonic partials. In this portion, every section records each occurrence of the fundamental frequency at the highest intensity compared with other partials. This is very similar to the decay portion. Table 7 shows the shift in frequency which occurs in each section within the sustain portion. These results indicate that most sections within the sustain portion focus on the frequency of 441.43066 Hz. There are only two sections with a fundamental frequency that focuses on another frequency: section S4 and S13. In these two sections, the fundamental frequency focuses on 442.10358 Hz.

The results within the sustain portion show that most partials begin to evolve throughout this portion. This is especially marked at section S13 where only the first nine partials occur. Table 8, which shows changes in the relative amplitude of the sound spectrum over time, clearly show the intensity of high frequencies [beginning at the seventeenth harmonic] becoming so low that their relative amplitudes become insignificant. In this portion, as all partials in the sound spectrum start to evolve, there are no partials with consistent relative amplitudes other than the high frequencies [beginning at the seventeenth harmonic] whose relative amplitudes need no longer remain under consideration.

The Release Portion

In this portion, the partials that appear in the spectrum consist of both harmonic and inharmonic partials. Most of the inharmonic partials occur after the fourth harmonic. Table 9 shows the changes in the relative amplitude of the sound spectrum over time within the release portion. After the first section, the peak amplitude displaces to the second harmonic. In the release portion, all the partials are evolving, and in the final two sections [R7 and R8], there are only three partials remaining. Table 10 shows the shift in frequency that occurs within the release portion. These results indicate that all sections within the release portion do not focus on a specific frequency. The fundamental frequencies shift very frequently among the frequencies 442.10358 Hz, 441.43066 Hz, 440.75775 Hz, 438.0661 Hz, 437.39319 Hz and 435.37445 Hz.

Resynthesis Process Results

In the process of resynthesis, each of the relative amplitude values obtained from the spectrum analysis are inserted into the equation (1) above according to their correlated time values and harmonic values. Finally all series of arrays y1, y2, y3,…, are concatenated into a double arrayed matrix and generated to produce the resynthesised sound.

DISCUSSION

There are five major findings obtained from this research. Firstly, the micro variations in the power spectrum of each of the partials are discussed. This is followed by a discussion on the shifting of the fundamental frequency, which was observed throughout the sections. The weak fundamental in the attack portion is discussed next, followed by the presence of inharmonicity in the release portion. Finally, data bandwidth in the generation of sound is considered.

Micro variations in the Sound Spectrum with respect to Time

The results show that the power spectrum of each of the partials varies with respect to time. These significant results indicate that spectrum changes over time are a very important feature in the synthesis of high quality or authentic sound. These changes in the loudness of each partial over time contribute to the timbre of the violin tone. These results are quite similar to findings by Mellody and Wakefield (1999), whose research focuses on the time-frequency characteristics of violin vibrato.

Shifting Fundamental Frequency

The shifting fundamental frequency that occurred in the sound spectrum of the recorded violin signal occurs in the sound spectrum of all the sections of the waveform (Table 11). The fundamental frequency does not focus on one specific value of frequency. It is most prominent in the attack portion and the release portion of the waveform.

During the recording process, the violin player was requested to play single notes, without the use of vibrato. It may be concluded therefore that the shifting frequency observed is not caused by the playing technique. As a further point to note, playing techniques such as vibrato, which consist of changing the frequency throughout the playing of a note, cause larger frequency fluctuations than the shifts obtained from this research. A possible cause of the frequency shifts observed in this research may be due to the inertia of the violin string. The greatest amount of shifting occurs in the attack and release portions of the waveform, when the string changes from a rest state to a vibrating state [attack portion] and again from a vibrating state back to a rest state [release portion]. The slight shifts in the decay and sustain portions are natural results of the uncertainty principle, which does not expect absolute and perfectly matching measurements in any observation but rather allow for a small margin of error.

These results indicate that the most frequent fundamental frequency is 441.43066 Hz, which is the focus point of most of the sections in the decay and sustain portions. Fundamental frequencies such as 440.75775 Hz, 438.0661 Hz, 437.39319 Hz, and 435.37445 Hz are only held by sections within the release portion. These frequencies are shown to be not accurate, as during resynthesis process, if such frequencies are selected for insertion into the resynthesis equations, the sound generated becomes not only artificial but the pitch of the synthesis is changed because of the inaccuracy.

Weak Fundamental Harmonics

A weaker strength or intensity of the fundamental harmonic, especially within the initial attack portion, was discovered. Table 3 indicates that within the first three windows, which occur from the starting point [0.00 seconds] to 0.06 seconds, the highest spectrum energy does not focus on the fundamental frequency, but instead focuses on the second harmonic. These results indicate that the intensities of the fundamental harmonic only dominate when the timbre of the tone becomes much stronger and more stable.

Inharmonicity

Within the release portion, several inharmonic overtones are found to occur. The inharmonic overtones only occur in the upper partials of the sound spectrum. This may be due to the mechanical stiffness as explained by Martin (1998). According to his explanation, freely vibrating strings may produce inharmonic partials. Further study is needed to explain the occurrence of the inharmonic partials.

Data Reduction

The amount of hard disk space required to store the originally recorded violin sample is 221 kilobytes. The hard disk capacity required to store the control file which controls the resynthesised sound, is only 13.7 kilobytes. Although this research only looks into the spectrum of the original sound signal, the sound resynthesised from these parameters is quite close to the original.

This is indicated through the sound discrimination experiment that was carried out in this research. In this sound discrimination experiment, the resynthesised sound, the original recorded violin sound and four other synthesised violin tones produced by commercially available synthesis techniques were loaded as soundfonts [a data format that defines the information required by a computer to create musical notes or sound effects through wavetable synthesis]. The release portion of the experimentally produced resynthesised sound [with its transient noise] was not included in the model due to the reasons explained in the earlier paragraphs. All the various violin tones were played back through a Creative Sound Blaster Live card. The other synthesised violin sounds were XWave PCI Audio OPL2/OPL3 Device, Yamaha SXG 50 Driver, Creative S/W Synth and the original chipset of the Creative Sound Blaster Live card. 20 listeners were then asked to discriminate between the six versions of the sound, which were played back twice in a set order. The listeners were asked to rate the realism of the played back sound on a five-point scale ranging from 0-20%, 20-40%, 40-60%, 60-80% and 80-100%. Results from this experiment indicated that about 90 percent of the listeners claimed that the synthesised sound from this research [which was obtained through time-varying spectral modelling technique] sounded similar to the original recorded violin sound (which was used for sound analysis in this research). However, the resynthesis process reduced the required hard disk capacity by about 93.8 percent. By reducing the usage of the hard disk capacity, the execution time of the sound resynthesis process may be increased. In addition, it may also enable a computer to store a whole range of a violin sound incorporating different techniques of playing.

CONCLUSION

It has been shown that for the single violin tone, the microvariations in the sound spectrum contribute significantly towards the timbre of the tone. When these parameters are used for resynthesis purposes, a significant file size saving is obtained, with minimal concessions to the sound quality. Further study needs to be done to complete the model for the violin for all notes within its range and for all techniques of performance. Further to this, detailed study needs to be done on further data simplification that may be undertaken without sacrificing the sound quality

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