ASPECTS IN MODELING AND REAL-TIME SYNTHESIS OF THE ACOUSTIC GUITAR

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Abstract

This paper will address the problem of modeling the acoustic guitar for real-time synthesis on signal processors. We will present a scheme for modeling the string for high-quality sound synthesis when the length of the string is changing dynamically. We will focus also on the problem of modeling the body of the guitar for real-time synthesis. Filter-based approaches were experimented by LPC estimation, IIR-filter synthesis and FIR-filter approximation. Perceptual evaluation was used and taken into account. Real-time synthesis was implemented on the TMS320C30 floating-point signal processor. The presentation includes audio examples.

Introduction

Computational modeling of musical instruments is an alternative to commonly used and more straightforward sound synthesis techniques like FM synthesis and waveform sampling. The traditional approach to efficient modeling of a vibrating string has been to use proper digital filters or transmission lines, see e.g. Karplus and Strong [1] and its extensions by Jaffe and Smith [2]. These represent "semiphysical" modeling where only some of the most fundamental features of the string, especially the transmission line property, are retained to achieve efficient computation. More complete finite element models and other kinds of physical modeling may lead to very realistic sounds but tend to be computationally too expensive for real-time purposes.

Modeling of the guitar body for real-time sound synthesis seems too difficult unless a digital filter approach to approximate the transfer function is used. The derivation of the detailed transfer function from mechanical and acoustical parameters seems impossible. The remaining choice is to estimate the transfer function filter from measurements of a real guitar or to design a filter that approximates the general properties of the real guitar body. In addition to strings and body the interactions between them (at least between the strings) should be included.

String Modeling

The natural way of modeling a guitar string is to describe it as a two-directional transmission or delay line (see Fig. 1a.) where the vibrational waves travel in both directions, reflecting at both ends. If all losses and other nonidealities are reduced to the reflection filters at the end points the computation of the ideal string is efficient by using two delay lines. The next problem is how to approximate the fractional part of the delay to achieve any (non-integer) length of the delay line. Allpass filters [2] are considered as a good solution if the string length is fixed. If the length is dynamically varying, however, it is very difficult to avoid transients and glitches when the integer part of the delay line must change its length.

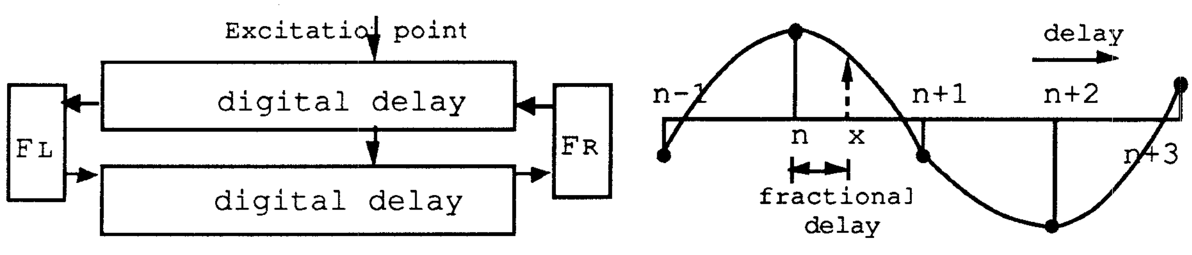

Fig. 1. a) Transmission-line principle of a string (left) and b) Lagrange interpolation for fractional delay approximation (right).

We solved this problem by using a FIR type Lagrange interpolator for fractional delay approximation [3,4,5], see Fig. 1b. It gives a good approximation of the desired delay properties and the amplitude response was found to be close enough to the ideal one. It was possible to implement a six string real-time synthesis model running on the TMS320C30 signal processor (sampling frequency 22 kHz). More details of this study are described in ref. [5]

Body modeling

The guitar body is a very complex resonator system that is mainly responsible for radiating efficiently the string vibration. The transfer function from the bridge to the listener can be measured approximately by exciting the bridge with a sharp impulse-like hit and by registering the radiated sound. An analysis shows that the envelope spectrum of the body response of a good guitar is relatively flat but there is a very large number of resonances starting from the fundamental Helmholtz resonance. From the perceptual point of view, the temporal envelope of the impulse response ("reverberation") is important, not only the detailed frequency domain response.

We studied the following methods of approximating the body transfer function for a real-time synthesis model:
- *Linear Prediction Analysis* (LPC) of a real guitar for estimating an all-pole filter of proper number of poles,
- *FIR filter approximation* of a real guitar for achieving useful quality,
- *IIR filter synthesis* to approximate the general frequency domain and time domain properties.

Perceptual criteria were found to be very important both for the evaluation of the methods and as intuition for the design of the filter models.

LPC analysis is a method to estimate the (impulse) response of a linear system by an all-pole filter. The transfer function of the guitar body is by no means of all-pole (minimum phase) character. Yet it seems reasonable to try LPC modeling due to its simplicity. A visual inspection of the Fourier spectrum of the body impulse response suggests that a minimum of about 20 resonances (pole pairs) per 1 kHz are needed to approximate the major resonance structure. This means a minimum of 200 resonances or 400 poles (as well as 400 coefficients) to model the body by a direct form IIR filter.

We found out that a better criterium can be found when comparing the temporal envelopes of the impulse responses, the original vs. the modeled one. The acoustic guitar we were modeling had a decay time constant of approximately 30 ms for the impulse response envelope. If the order of LPC is not high enough the impulse response will remain shorter than the original due to underestimated Q-values of the resonances. This is important from the perceptual point of view. In our experiments we found out that LPC of order 500 worked well in this sense for a 22 kHz sampling frequency. This means a filter complexity that is within the reach of a single TMS320C30 floating point processor.

Using **FIR** filters is a straightforward approach if the truncated impulse response, when used as coefficients for the body filter, leads to an acceptable result and to a real-time realization. Based on perceptual evaluation we found out that a minimum of 30 ms (preferably 60 ms) of the impulse response of the real guitar body must be used. This is again within the reach of a single TMS320C30 if a sampling frequency of 22 kHz is used.

IIR filters tend to be more efficient than FIR structures if there exists a method to design an optimized filter. We tried out several approaches to approximate the general structure of the body response properties by IIR filters. The resonances of the model filter should follow the principle of "ordered randomness". A simple technique is to design a parallel connection of second order sections, each one implementing a single resonance. The resonance frequences must be randomized but otherwise uniformly spaced and the resonance amplitudes should be nearly equal. Due to the unefficient computation of such filters on the TMS320C30 the order of the filter should be about 100 or lower for real-time implementations. To achieve this we designed a filter where the resonance density decreased for higher frequencies, following the perceptual rule of uniform number of resonances for each critical band (Bark unit). The results are useful although not as natural as from the two first estimation methods.

The experiments show that relatively detailed and natural sounding model-based synthesis of the acoustic guitar is possible by a pair of fast signal processors and on a single DSP chip in the near future.

References

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